Bose®
Panaray® MA12 modular line array loudspeaker

BOSE PROFESSIONAL SYSTEMS DIVISION – COMPLETE COMMERCIAL SOLUTIONS

Bose® Panaray® MA12 modular line array loudspeaker

TECHNICAL INFORMATION

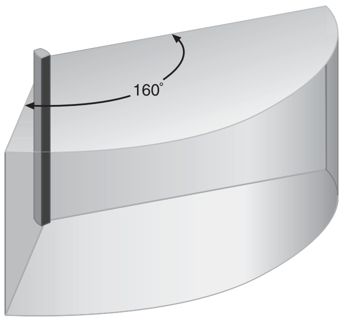
Maximum acoustic output[1]
113dB-SPL @ 1m
A single MA12 produces cylindrical waves above 2kHz. Please be careful extrapolating SPL at greater distances, as a true cylindrical source falls off at 3dB per doubling of distance, and a conventional spherical source falls off at 6dB per doubling of distance (inverse square law).

Sensitivity[2]
88dB-SPL @ 1W, 1m

Frequency range
155Hz to 12kHz (-3dB)
100Hz to 16kHz (-10dB)

Dispersion (–6dB point, average, 1-4kHz)
Horizontal 160°
Vertical coverage is a band of constant height as shown above

REPRESENTATION OF DISPERSION



1 - 4kHz (-6dB point average)

Long-term power handling[3]
300W continuous

Impedance
8Ω nominal

Mechanical specifications
Dimensions:
5.1"D x 4.2"W x 38.7"H
(13 cm x 10.7 cm x 98.4 cm)
Weight: 20 lb (9 kg)
Available colors: white and black
Connectors: two parallel-wired NL4 connectors and one two-position barrier strip
Enclosure construction: aluminum

GENERAL DESCRIPTION

The Panaray MA12 modular line array is a 300W loudspeaker. As a line array element, this product is designed for use in singles and multiples in permanently-installed indoor applications. The slim profile of the MA12 speaker provides an attractive design element. As an addition to the Bose line of Panaray loudspeakers, this product combines innovative design technology with Bose standards for acoustic quality and performance.
The MA12 speaker offers the following features:

- One-way system for accurate voice and light music reproduction
- Full-range system performance when combined with a Bose bass loudspeaker
- Line array characteristics that put the sound where you want it:
    - Radiation is a band of sound at a constant height, to significantly reduce ceiling and floor reflections

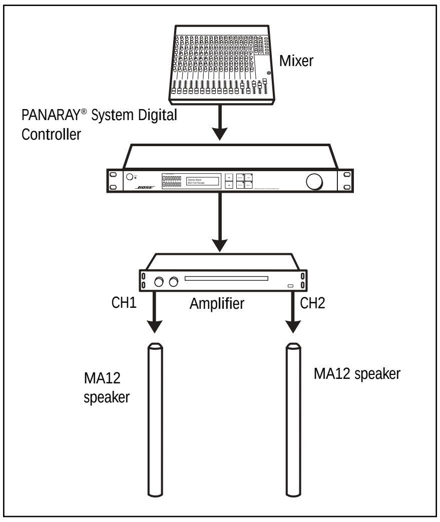
---

[1] Full bandwidth pink noise is applied to the MA12 speaker and amplified to a level at the loudspeaker terminals corresponding to the long-term rated power handling of the speaker. The average sound pressure level (dB-SPL) is measured at distances from the speaker in an anechoic environment.

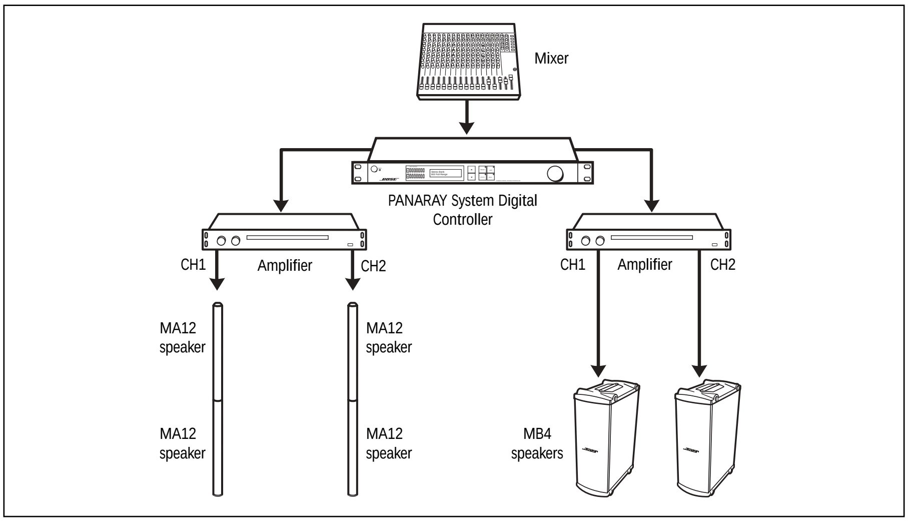
[2] Full bandwidth pink noise is applied to the MA12 speaker and amplified to a level at the loudspeaker terminals corresponding to1 Watt as referenced to the nominal impedance. The average sound pressure level (dB-SPL) is measured at 1 meter from the speaker in an anechoic environment.

[3] Full bandwidth noise, meeting the IEC Standard #268-5 is applied to the MA12 speaker and amplified to a level at the loudspeaker terminals corresponding to the power handling of the loudspeaker. The loudspeaker must show no visible damage or measurable loss of performance after 100 hours of continuous testing.

BASIC VOICE SYSTEM



EXPANDED FULL-RANGE SYSTEM



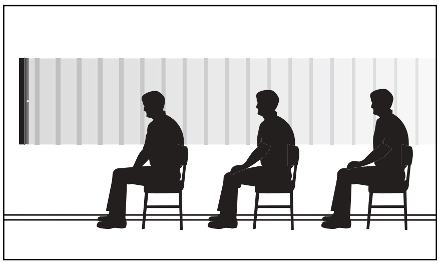
  - Wide horizontal dispersion for broad coverage
  - Longer throw than conventional sources with less SPL variation.
    This means you will experience sound that is not too loud up front, and not too soft far away.
- Low sensitivity to microphone feedback
- Slender, aesthetic styling in white or black
- A full complement of brackets available for product installations

MA12 loudspeaker modules are used with the Panaray system digital controller or the Bose amplifiers in combination with the MA12 speaker plug-in EQ card.
Active equalisation assures smooth, accurate spectral response across the entire operating range of the module. Use of the Panaray system digital controller or amplifier EQ card also allows for crossover of the MA12 speaker with a Bose bass loudspeaker.

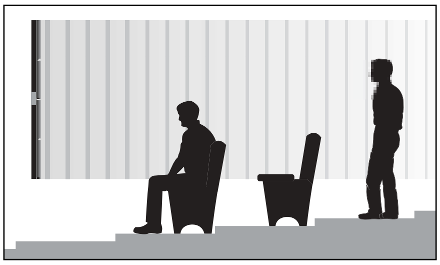
INSTALLATION

Installation and use instructions are included with the loudspeaker.

The MA12 modular line array speakers are designed to be wall or floor mounted at the listener's ear height.



MA12 speakers can be installed end-on-end to create a taller line array. The height of the line array must correspond to the ear height range of the listeners, in sitting and standing positions, in the installation venue.



LOUDSPEAKER CONFIGURATION

The Bose MA12 loudspeaker module can be used with the Bose M2150, 1600 and 1800 amplifiers. It is also compatible with industry standard amplifiers rated for 4 to 8Ω loads. The MA12 loudspeaker is packaged one per carton.

ENGINEERS' AND ARCHITECTS' SPECIFICATIONS

The MA12 speaker shall be a multiple driver, full-range system with matched active equalisation as follows:

The transducer complement shall consist of twelve 2.2" (5.7 cm) full-range drivers mounted in a vertical line array. Each driver shall have a rated impedance of 3.2Ω and be wired in series/parallel, resulting in a composite nominal impedance of 8Ω.

The nominal horizontal beamwidth of the loudspeaker shall be 160°, and the vertical coverage is a band of constant height.
The power handling capacity of the loudspeaker shall be 300W continuous pink noise, band-limited from 155Hz to 12kHz (-3dB).

The loudspeaker shall be provided with a ducted vent system, tuned at 100Hz.
The input connector of the loudspeaker module shall consist of two parallel-wired Speakon® NL4 receptacles and one two-position barrier strip.

The enclosure of the MA12 speaker module shall be aluminum. Its outer dimensions shall be 5.1"D x 4.2"W x 38.7"H (13 cm x 10.7 cm x 98.4 cm). Its weight shall be 20 lb (9kg).

The loudspeaker system shall be supplied with a separate system controller, to be connected before the input(s) of the system power amplifier(s), or with an MA12 EQ card, to be installed in a Bose M2150, 1600 or 1800 amplifier.

The loudspeaker shall be the Bose Panaray MA12 modular line array loudspeaker.

PANARAY MA12 ACCESSORIES

The following accessories are available from Bose:
- PC028333 - WB-MA12 wall mount bracket (0°-10° pitch) – black
- PC028334 - WB-MA12 wall mount bracket (0°-10° pitch) – white
- PC028673 - WBP-MA12 wall mount bracket (0°-10° pitch, +/- 90° yaw) – black
- PC028674 - WBP-MA12 wall mount bracket (0°-10° pitch, +/- 90° yaw) – white
- PC028337 - CB-MA12 coupling bracket – black
- PC028338 - CB-MA12 coupling bracket – white
- PC030114 - PSA-12 stand adapter – black
- PC029388 - CVT-12 70V/100V transformer – black
- PC029389 - CVT-12 70V/100V transformer – white

PANARAY® MA12 SPEAKER DATA

For additional information on the MA12 including polar data please refer to the following documents:

(1) Panaray MA12 modular line array: technical information

(2) Panaray MA12 modular line array: technical foundation and discussion

These documents can be downloaded from the Bose® professional systems website at http://pro.bose.com.

SAFETY AND REGULATORY COMPLIANCE

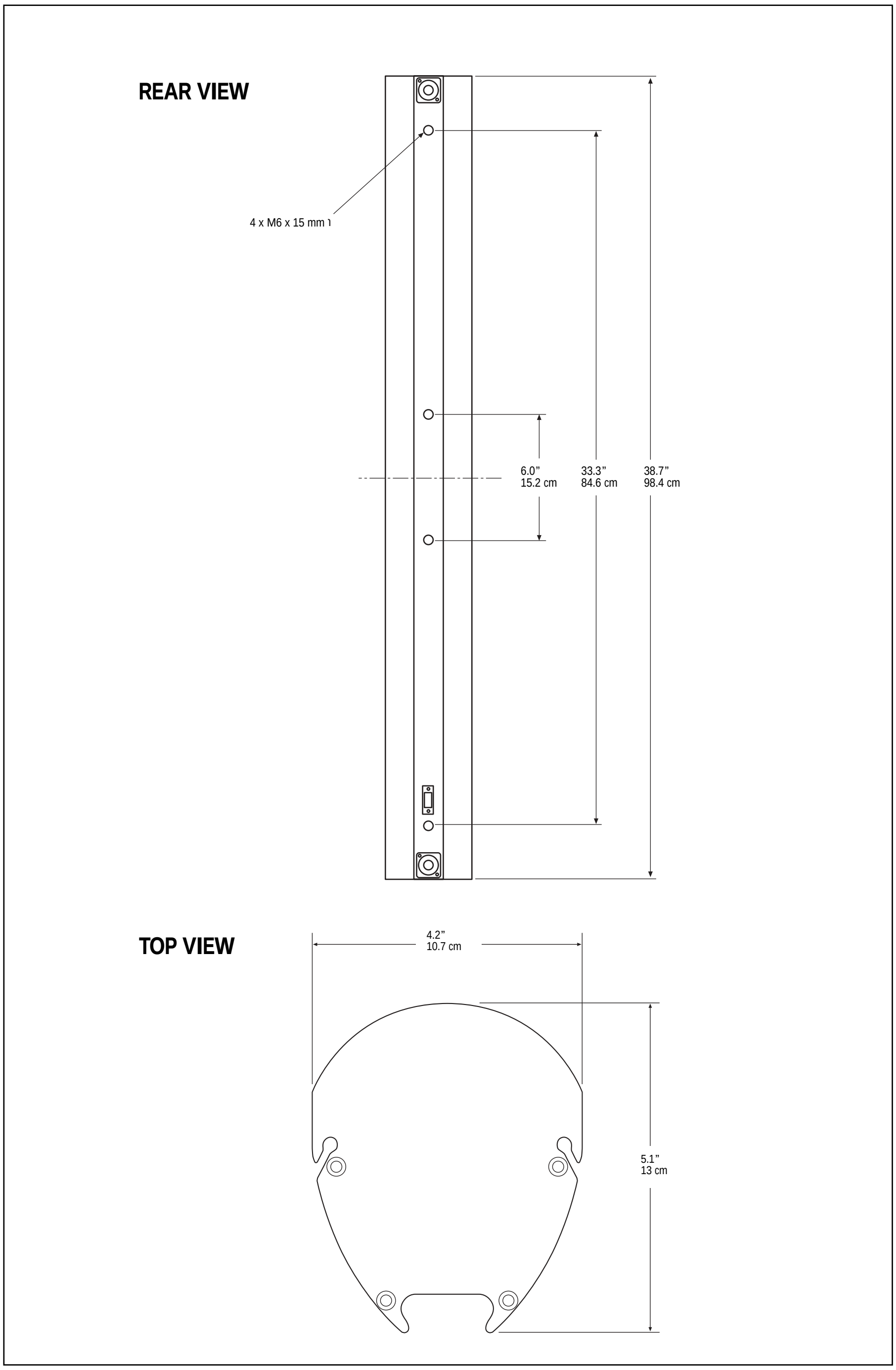
The Bose MA12 loudspeaker complies with ANSI/EIA 636, recommended loudspeaker safety practices and EMC directive 89/336/EEC and article 10 (1) of the directive in compliance with EN50081-1, EN50082-1, as signified by the CE mark.

WARRANTY INFORMATION

The Bose MA12 loudspeaker is covered by a 5-year, transferable limited warranty.

Subject to change without notice.

DIMENSIONS



Better sound through research®

Products and accessories

Bose®
Panaray® MB 4 modular bass loudspeaker

BOSE PROFESSIONAL SYSTEMS DIVISION – COMPLETE COMMERCIAL SOLUTIONS

Customer Focus | Acoustic design | Listen before you buy | Project management | Products and accessories | Installation services | After sales services

Bose® Panaray® MB4 modular bass loudspeaker

TECHNICAL INFORMATION

Frequency Range
40Hz to 300Hz ±3dB

Sensitivity[1]
87dB-SPL @ 1W, 1m (pink noise)

Maximum Acoustic Output[2]
110dB-SPL, @ 1m (pink noise)

Dispersion
Omnidirectional at all frequencies below 200Hz

Long-Term Power Handling[3]
200W continuous

Impedance
8Ω nominal

MECHANICAL SPECIFICATIONS

Dimensions: 18.1"D x 10.2"W x 26.7"H (46 cm x 25.9 cm x 67.8 cm)

Weight: 45 lb (20 kg)

Connectors: Two (2) parallel-wired NL4 connectors

Available colors: White and Black

Enclosure construction:
High-density MultiFiber multi-layer wood particle panel with high-impact foamed polypropylene end caps.

PANARAY MB4 LOUDSPEAKER SIZING

Incremental bass output can be achieved by combining multiple MB4 units in close proximity.

# of Panaray MB4s	Max Output (dB)
1	110
2	116
3	119
4	122

A single MB4 speaker is well matched in output and bandwidth to one or two Panaray 402-II, 502A or MA12 loudspeakers, providing a full-range solution for distributed or portable systems. Two or three MB4 speakers are a good match for higher-output loudspeakers, like the Panaray 802®-III speaker, for engineered or portable systems.

For larger spaces that require even higher SPL, four or more MB4 speakers can be combined with each Panaray LT series loudspeaker.

GENERAL DESCRIPTION

The Panaray MB4 modular bass loudspeaker is a 200W, low-frequency loudspeaker that can be used for indoor permanent installations and for portable applications. Multiple MB4 speakers can be combined to effectively create a larger single bass enclosure resulting in higher SPL.

The MB4 speaker's output, bandwidth and size make it useful as a single bass unit, or as a modular building block to match other Bose® products in most professional systems.

The MB4 speaker can also be used in conjunction with the MB4 endfire or broadside bass array brackets and the Panaray system digital controller to create directional bass.

The Panaray MB4 modular bass loudspeaker offers the following features:

- Wide frequency range, from 40Hz to 300Hz, which can be easily matched to any Bose professional loudspeaker
- High impact output for dominant bass material
- Rugged portable enclosure with metal grille and carrying handles
- Distinctive styling
- Recessed back plate with two Speakon® NL4 connectors

---

[1] Band-limited pink noise is applied to the MB4 speaker and amplified to a level at the loudspeaker terminals corresponding to 1 Watt as referenced to the nominal impedance. The average sound pressure level (dB-SPL) is measured at 1 metre from the speaker in an anechoic environment.

[2] Band-limited pink noise is applied to the MB4 speaker and amplified to a level at the loudspeaker terminals corresponding to the long-term rated power handling of the speaker. The average sound pressure level (dB-SPL) is measured at 1 metre from the speaker in an anechoic environment.

[3] Band-limited noise, meeting the IEC Standard #268-5 is applied to the MB4 speaker and amplified to a level at the loudspeaker terminals corresponding to the power handling of the loudspeaker. The loudspeaker must show no visible damage or measurable loss of performance after 100 hours of continuous testing.

- Full complement of brackets to simplify installations

The MB4 speaker can be used with the Bose Panaray system digital controller to provide smooth accurate spectral response over the entire operating range of the system.

LOUDSPEAKER CONFIGURATION

The MB4 modular bass loudspeaker can be used with the Bose® 1600 and 1800 amplifiers. The MB4 plug-in EQ card must be used with these amplifiers to provide active equalisation for the system.

When used with the Panaray® system digital controller, the MB4 speaker is also compatible with industry standard amplifiers rated for 4Ω to 8Ω loads.

The MB4 speaker is packaged one per carton.

INSTALLATION

Installation and user instructions are included with the loudspeaker.

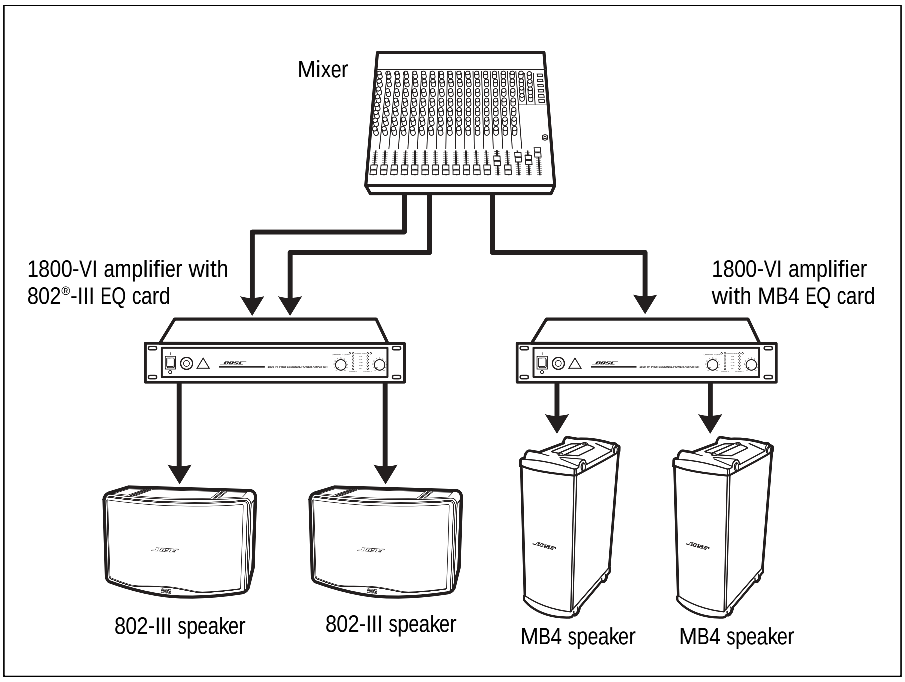
MB4 modular bass loudspeakers are designed to be wall mounted. They can also be assembled into bass arrays, by using the MB4 endfire or broadside brackets in conjunction with the Bose Panaray system digital controller.

No stands are required for portable use.

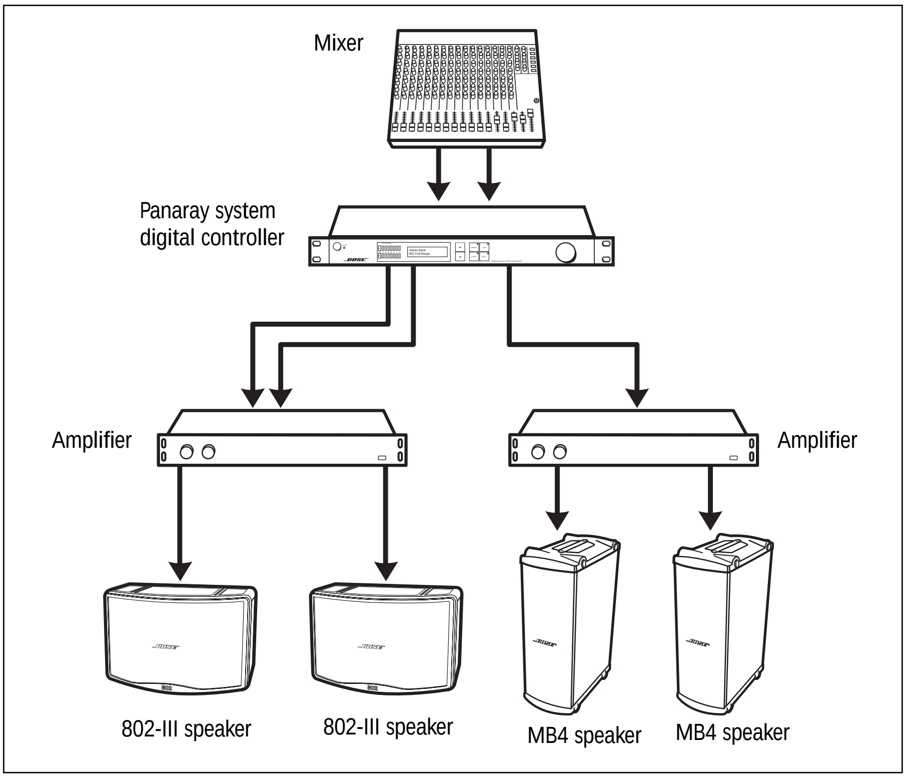
ACCESSORIES

The following versatile accessories are available from Bose:
- PC027056 - MB4 single-unit wall mount bracket - black, PC027057 - white
- PC028664 - MB4 endfire 2x bass array bracket
- PC028665 - MB4 endfire 4x bass array bracket
- PC028663 - MB4 broadside 2x bass array bracket
- PC030132 - MB4 Protection Bag

FULL-RANGE SYSTEM WITH BOSE® AMPLIFERS


Mixer
1800-VI amplifier with 802®-III EQ card
1800-VI amplifier with MB4 EQ card
802-III speaker
802-III speaker
MB4 speaker
MB4 speaker

FULL-RANGE SYSTEM WITH PANARAY® SYSTEM DIGITAL CONTROLLER



ENGINEERS' AND ARCHITECTS' SPECIFICATIONS

The loudspeaker shall be a multiple driver, low-frequency system with matched active equalisation as follows:

The transducer complement shall consist of four low-frequency drivers of 5.2" (13.3 cm) diameter, mounted symmetrically in vertical pairs. Each driver shall have rated impedance of 8Ω and shall be wired in series/parallel, resulting in composite nominal impedance of 8Ω.

The loudspeaker system sensitivity shall be 87dB-SPL in the 40Hz to 300Hz frequency range with measurements referenced to 1W (2.85V) pink noise input at 1m.
The dispersion shall be omnidirectional at all frequencies below 200Hz. The power handling capacity of the loudspeaker shall be 200W continuous pink noise, band-limited from 40Hz to 300Hz. The loudspeaker shall be a vented enclosure, tuned at 45Hz.

The input connectors of the Panaray MB4 speaker shall consist of two parallel Speakon® NL4 receptacles.

The enclosure of the MB4 loudspeaker shall be composed of high-density MultiFiber multi-layer wood particle panel with high-impact foamed polypropylene end caps.

Its outer dimensions shall be 18.1"D x 10.2"W x 26.7"H (45 cm x 25.9 cm x 67.8 cm). Its weight shall be 45 lb (20 kg).

The loudspeaker shall be the Panaray® MB4 modular bass loudspeaker.

The loudspeaker system shall be used with a separate system controller, to be connected to the input(s) of the system power amplifier(s), or with an MB4 EQ card, to be installed in a Bose 1600 or 1800 amplifier. The Panaray system digital controller shall provide active electronic equalisation and crossover functions.

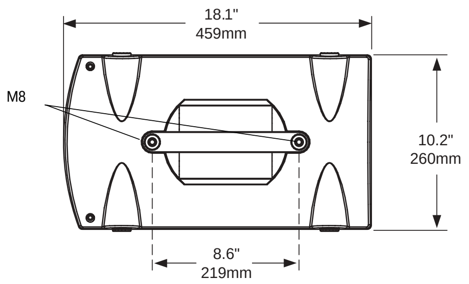
SAFETY AND REGULATORY COMPLIANCE

All versions of the Panaray MB4 modular bass loudspeakers comply with ANSI/EIA 636, Recommended Loudspeaker Safety Practices and EMC Directive 89/336/EEC and Article 10 (1) of the Directive in compliance with EN50081-1, EN 50082-1, as signified by the CE mark.

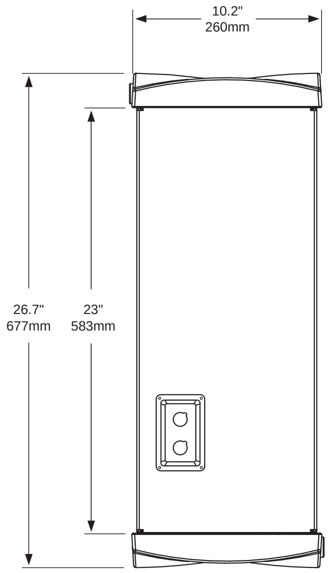
WARRANTY INFORMATION

All versions of the Bose Panaray MB4 modular bass loudspeakers are covered by a 5-year, transferable limited warranty.

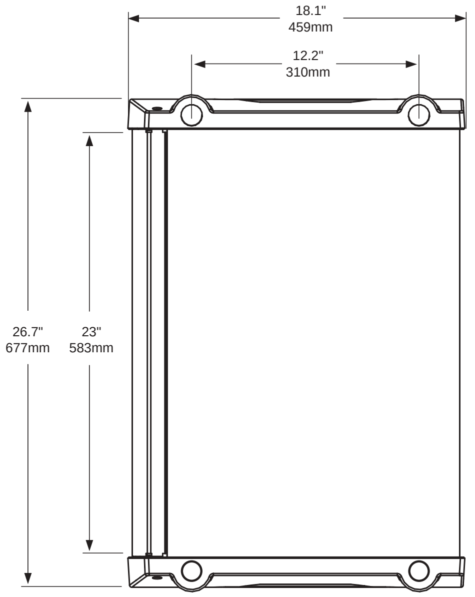
DIMENSIONS



TOP VIEW



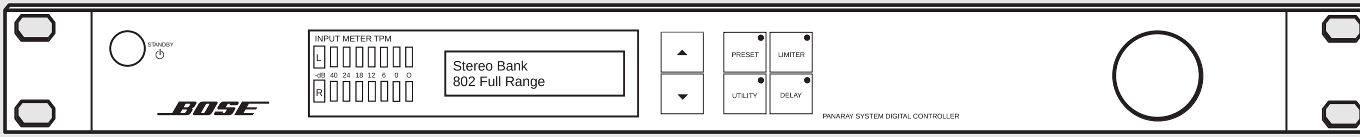
BACK VIEW



SIDE VIEW

PANARAY® System Digital Controller

Installation guide


STANDBY
INPUT METER TPM
L
-dB 40 24 18 12 6 0 O
R
Stereo Bank
802 Full Range
PRESET
LIMITER
UTILITY
DELAY
BOSE
PANARAY SYSTEM DIGITAL CONTROLLER

pro.bose.com

Table of Contents

Safety Information 4

1.0 Introduction 8
2.0 Front Panel 9
3.0 Rear Panel 10
4.0 Signal Flow 11
  4.1 Ch1 and Ch2 Inputs 12
  4.2 EQ and Crossover 13
  4.3 2 x 4 Router 14
  4.4 Delay and Limiter 15
5.0 PRESET Button. 16
  5.1 Stereo Bank 17
    5.1.1 Stereo Mid-High with Mono Bass 18
  5.2 Mono Bank 19
  5.3 Bass Array Bank 20
  5.4 Dual Mono Operation 22
6.0 DELAY Button 24
7.0 LIMITER Button. 25
8.0 UTILITY Button 26
  8.1 User Lockout 28
9.0 Router Functionality. 29
10.0 Optimizing Signal-to-Noise. 31
11.0 Custom Preset Mode. 33
12.0 Bass Array Presets 34
  12.1 Bass Array Setup and Connections 35
13.0 Bose® Speaker EQ Preset List 38
  13.1 Mono Bank 39
  13.2 Stereo Bank 46
  13.3 Bass Array Bank 53
14.0 Technical Information. 54

Appendix: PANARAY® System Digital Controller Quick-Reference Guide. 55

Important Safety Instructions

Warning (English)

To reduce the risk of fire or electric shock, do not expose the unit to rain or moisture.

CAUTION

RISK OF ELECTRICAL SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK,
DO NOT REMOVE COVER (OR BACK).
NO USER-SERVICEABLE PARTS INSIDE.
REFER SERVICING TO QUALIFIED PERSONNEL.

These CAUTION marks may be located on the back of the PANARAY® System Digital Controller.

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated dangerous voltage within the system enclosure that may be of sufficient magnitude to constitute a risk of electric shock.

The exclamation point within an equilateral triangle, as marked on the system, is intended to alert the user to the presence of important operating and maintenance instructions in this owner's guide.

Important Safety Instructions

1. Read and follow these instructions.
2. Heed all warnings–on the product and in the owner's guide.
3. Do not use this apparatus near water or moisture.
4. Clean only with a dry cloth.
5. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions. To ensure reliable operation of the product and to protect it from overheating, put the product in a position and location that will not interfere with its proper ventilation.
6. Do not install near any heat sources, such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
7. Do not defeat the safety purpose of the grounding-type plug. A grounding-type plug has two blades and a third grounding prong. The third prong is provided for your safety. If the provided plug does not fit in your outlet, consult an electrician for replacement of the outlet.
8. Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
9. Only use attachments/accessories specified by the manufacturer.
10. Use only with a cart, stand, tripod, bracket or table specified by the manufacturer or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
11. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way: such as power supply cord or plug is damaged; liquid has been spilled or objects have fallen into the apparatus; the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped–Do not attempt to service this product yourself. Opening or removing covers may expose you to dangerous voltages or other hazards. Please call Bose® to be referred to an authorized service center near you.
12. To prevent risk of fire or electric shock, avoid overloading wall outlets, extension cords, or integral convenience receptacles.
13. Do not let objects or liquids enter the product– as they may touch dangerous voltage points or short-out parts that could result in a fire or electric shock.
14. See product enclosure back for safety related markings.
15. No naked flame sources, such as lighted candles, should be placed on the apparatus.
16. Product must be rack-mounted and installed in accordance with local building codes.
17. Product is not intended for marine applications.

Advarsel (Dansk)

Enheden må ikke udsættes for regn eller fugtighed af hensyn til risikoen for brand eller elektrisk stød.

FORSIGTIGHED

RISIKO FOR ELEKTRISK STØD
MÅ IKKE ÅBNES

FJERN IKKE LÅGET (ELLER BAGKLÆDNINGEN)
AF HENSYN TIL RISIKOEN FOR ELEKTRISK STØD.
DER ER INGEN DELE INDEN I, DER KAN SERVICERES
AF BRUGEREN. OVERLAD SERVICERINGEN
TIL KVALIFICERET PERSONALE.

Disse FORSIGTIGHEDSMÆRKER er anbragt på de PANARAY® System Digital Controller.

Symbolet med et lyn med pilespids i en ligesidet trekant skal gøre brugeren opmærksom på tilstedeværelsen af en ikke-isoleret farlig spænding i det lukkede system, der kan være tilstrækkelig kraftig til at give elektrisk stød.

Udråbstegnet i en ligesidet trekant, som systemet er mærket med, er beregnet til at advare brugeren om vigtige instruktioner til betjening og vedligeholdelse i denne betjeningsvejledning.

Vigtige sikkerhedsinstruktioner

1. Læs og følg disse instruktioner.
2. Læg mærke til alle advarsler - på produktet og i betjeningsvejledningen.
3. Anvend ikke dette apparat i nærheden af vand eller fugtighed.
4. Rengør kun med en tør klud.
5. Undgå at blokere ventilationsåbningerne. Installér i overensstemmelse med producentens instruktioner - Til sikring af pålidelig funktion og beskyttelse mod overophedning skal produktet anbringes i en position og på et sted, der ikke hindrer korrekt ventilation.
6. Installér ikke i nærheden af varmekilder, såsom radiatorer, varmeovne, komfurer eller andre apparater (herunder forstærkere) der frembringer varme.
7. Omgå ikke sikkerheden ved brug af et ikke jordforbundet stik. Et jordforbundet stik har to ben og et tredje jordben. Det tredje ben er der af hensyn til din sikkerhed. Hvis det medfølgende stik ikke passer i stikkontakten, skal du kontakte en elektriker for at få stikkontakten udskiftet.
8. Beskyt netledningen mod at blive trådt på eller klemt, specielt ved stik, stikkontakter og der, hvor den kommer ud af apparatet.
9. Brug kun tilslutningsudstyr/tilbehør, der er angivet af producenten.
10. Brug det kun sammen med en vogn, eller bord, der er angivet af producenten eller solgt sammen med apparatet. Når der anvendes en vogn, skal der udvises forsigtighed ved kørsel med vogn/apparat, så det ikke vælter.
11. Overlad al serviceringen til kvalificeret servicepersonale. Service er påkrævet, hvis apparatet på nogen måde er blevet beskadiget: f.eks. hvis netledningen eller stikket er blevet beskadiget, der er blevet spildt væske eller faldet genstande ned i apparatet, apparatet er blevet udsat for regn eller fugt, hvis det ikke fungerer normalt, eller hvis det er blevet tabt - forsøg ikke at reparere apparatet selv. Åbning eller fjernelse af låget kan udsætte dig for farlige spændinger eller andre farer. Kontakt Bose® for at få oplysninger om det nærmeste servicecenter.
12. Undgå at overbelaste stikkontakter, forlængerledninger eller indbyggede stik af hensyn til risikoen for brand eller elektrisk stød.
13. Undgå at genstande eller væsker kommer ind i produktet, da de kan berøre steder med farlige spændinger eller kortslutte dele, hvilket kan resultere i brand eller elektrisk stød.
14. Se på bagklædningen angående mærkater, der er relevante for sikkerheden.
15. Der må ikke anbringes åben ild såsom tændte lys på apparatet.
16. Produktet skal monteres i rack og installeres i henhold til lokale bygningsregler.
17. Produktet er ikke beregnet til marineprogrammer.

Important Safety Instructions

Vorsicht! (Deutsch)

Dieses Gerät darf weder Regen noch Feuchtigkeit ausgesetzt werden, damit Brände durch Kurzschluss oder elektrische Schläge verhindert werden.

VORSICHT!

STROMSCHLAGGEFAHR!
NICHT ÖFFNEN.

ACHTUNG! ZUR VERMEIDUNG VON ELEKTRISCHEN SCHLÄGEN WEDER DIE ABDECKUNG NOCH DIE RÜCKWAND ENTFERNEN. IM INNEREN BEFINDEN SICH KEINE VOM BENUTZER ZU WARTENDEN TEILE. WENDEN SIE SICH BEI NOTWENDIGEN REPARATUREN AN QUALIFIZIERTES FACHPERSONAL.

Diese Sicherheitshinweise befinden sich auf der Rückseite des PANARAY® System Digital-Controller.

Das Blitzsymbol mit Pfeilspitze in einem gleichseitigen Dreieck soll den Benutzer auf das Vorhandensein einer nichtisolierten gefährlichen elektrischen Spannung innerhalb des Systemgehäuses hinweisen, die so hoch ist, dass eine Gefährdung durch elektrische Schläge besteht.

Das Ausrufungszeichen in einem gleichseitigen Dreieck, wie es auf dem System angebracht ist, soll den Benutzer auf wichtige Bedienungs- und Wartungsanweisungen in dieser Bedienungsanleitung aufmerksam machen.

Wichtige Sicherheitshinweise

1. Lesen Sie die folgenden Sicherheitshinweise noch vor der Inbetriebnahme des Geräts aufmerksam durch.
2. Beachten Sie alle Warn- und Sicherheitshinweise am Gerät und in dieser Bedienungsanleitung.
3. Verwenden Sie dieses Gerät nicht in der Nähe von Wasser oder Feuchtigkeit.
4. Reinigen Sie das Gerät nur mit einem sauberen, trockenen Tuch.
5. Achten Sie darauf, dass die Lüftungsöffnungen nicht blockiert sind. Stellen Sie das Gerät gemäß den Herstelleranweisungen auf. Stellen Sie es nur an einem Ort auf, an dem eine gute Luftzirkulation gewährleistet ist, um den zuverlässigen Betrieb des Geräts sicherzustellen und es gegen Überhitzung zu schützen.
6. Stellen Sie das Gerät nicht in der Nähe von Wärmequellen auf, wie Heizkörpern, Wärmespeichern, Öfen oder anderen Geräten (auch Verstärkern), die Wärme erzeugen.
7. Beeinträchtigen Sie in keiner Weise die Schutzfunktion des Schutzkontaktsteckers. Ein Schutzkontaktstecker hat zwei Stromkontakte und einen dritten Erdungskontakt. Dieser Erdungskontakt dient Ihrer Sicherheit. Falls der mitgelieferte Schutzkontaktstecker nicht n Ihre Steckdose passt, wenden Sie sich an einen qualifizierten Elektriker, um die Steckdose auszutauschen.
8. Verlegen Sie das Netzkabel so, dass es keine Stolpergefahr darstellt und nicht beschädigt werden kann - insbesondere im Bereich von Steckern und Steckdosen und dort, wo das Netzkabel aus dem Gerät herausgeführt wird.
9. Verwenden Sie nur Zubehör-/Anbauteile, die vom Hersteller zugelassen sind.
10. Verwenden Sie für das Gerät nur Rollwagen, Ständer, Dreibeine, Halterungen oder Tische, die vom Hersteller zugelassen sind oder zusammen mit dem Gerät verkauft werden. Falls Sie einen Rollwagen verwenden, dürfen Sie die Einheit Gerät/Rollwagen nur mit Vorsicht bewegen, damit Verletzungen beim möglichen Umkippen ausgeschlossen sind.
11. Wenden Sie sich bei allen Reparatur- und Wartungsarbeiten nur an qualifiziertes Kundendienstpersonal. Eine Reparatur ist erforderlich, wenn das Gerät in irgendeiner Weise beschädigt wurde: Beispielsweise falls das Netzkabel oder der Netzstecker beschädigt ist, Flüssigkeit oder Fremdkörper in das Gerät gelangt sind oder das Gerät Regen bzw. Feuchtigkeit ausgesetzt war, nicht einwandfrei arbeitet oder heruntergefallen ist. Versuchen Sie in solchen Fällen keinesfalls, das Gerät selbst zu reparieren. Öffnen oder entfernen Sie unter keinen Umständen Gehäuseabdeckungen, da Sie andern falls mit gefährlichen elektrischen Spannungen in Berührung kommen oder anderen Gefahren ausgesetzt sein könnten. Wenden Sie sich telefonisch an Bose®, um die Anschrift eines autorisierten Kundendienstzentrums in Ihrer Nähe zu erfragen.
12. Vermeiden Sie Gefahren durch Brände oder elektrische Schläge, indem Sie Steckdosen, Verlängerungskabel und integrierte Gerätesteckdosen nicht überlasten.
13. Lassen Sie keine Flüssigkeiten oder Fremdkörper in das Gerät gelangen - sie können unter gefährliche elektrische Spannung gesetzt werden oder Bauteile kurzschließen und folglich Brände und elektrische Schläge auslösen.
14. Beachten Sie die Sicherheitshinweise auf der Geräterückseite.
15. Stellen Sie keine offenen Flammen, wie brennende Kerzen, auf das Gerät.
16. Rack-Einbau und Installation des Produkts müssen in Übereinstimmung mit den vor Ort geltenden Bauordnungen erfolgen.
17. Das Produkt ist nicht für den Einsatz in Wassernähe bestimmt.

Advertencia (Español)

Con el fin de reducir el riesgo de descarga eléctrica, no exponga la unidad a lluvia o humedad.

PRECAUCIÓN

RIESGO DE DESCARGA ELÉCTRICA
NO ABRIR

PRECAUCIÓN: PARA REDUCIR EL RIESGO DE DESCARGA ELÉCTRICA, NO QUITE LA CUBIERTA (O LA PARTE TRASERA). EL INTERIOR NO CONTIENE COMPONENTES REPARABLES POR EL USUARIO. PARA REPARACIONES, LLEVE EL APARATO A PERSONAL CUALIFICADO.

Estas marcas de PRECAUCIÓN podrian estar situadas en la parte trasera del Controlador Digital para Systemas PANARAY®

El símbolo de un relámpago con una flecha, dentro de un triángulo equilátero, avisa al usuario que dentro de la caja del sistema puede haber voltaje de una magnitud suficiente para constituir un riesgo de descarga eléctrica.

El signo de exclamación dentro de un triángulo equilátero, según las marcas en el sistema, sirven para avisar al usuario de que existen instrucciones de operación y mantenimiento importantes en esta guía del usuario.

Importantes instrucciones de seguridad

1. Lea y siga estas instrucciones.
2. Respete todos los avisos – del producto y de la guía del usuario.
3. No utilice este aparato cerca del agua o humedad.
4. Limpie sólo con un trapo seco.
5. No bloquee las aperturas de ventilación. Instale según las instrucciones del fabricante – Para garantizar que el producto funciona de manera fiable y con el fin de evitar que se sobrecaliente, coloque el producto en una posición y ubicación que no interfiera con su ventilación adecuada.
6. No lo instale cerca de fuentes de calor, tales como radiadores, salidas de aire caliente, hornos o demás aparatos (inclusive amplificadores), que generen calor.
7. No elimine el dispositivo de seguridad ofrecido por el enchufe con toma a tierra. Los enchufes con toma a tierra tienen dos patas para la corriente, y una tercera para la toma de tierra. La tercera pata está instalada a efectos de seguridad. Si el enchufe suministrado no encaja en sus tomas de electricidad, consulte con un electricista para sustituirlo.
8. Proteja el cable eléctrico de forma que nadie lo pise o quede apretado, particularmente cerca de los enchufes, receptáculos de tomas múltiples, y en el lugar en el que sale del aparato.
9. Utilice sólo conexiones y accesorios suministrados por el fabricante.
10. Utilice sólo con un carrito, soporte, trípode, abrazadera o mesa suministrados por el fabricante o vendidos junto al aparato. Cuando se utilice un carrito, tenga cuidado al mover la combinación de carrito/aparato para evitar lesiones en caso de que se caiga.
11. Toda reparación debe ser realizada por personal cualificado. Lleve el aparato a reparar si éste sufre cualquier daño, como por ejemplo: si el cable eléctrico o enchufe están dañados, se ha derramado líquido o han caído objetos dentro del aparato; el aparato ha sido expuesto a la lluvia o a la humedad, no funciona normalmente, o ha caído – No intente reparar este producto usted mismo. Abrir o sacar las tapas le expondrá a volta jes peligrosos o a otros peligros. Por favor llame a Bose® para que le indiquen un centro de reparación autorizado en su zona.
12. Para impedir el riesgo de descarga eléctrica, evite la sobrecarga de los enchufes, cables de extensión o receptáculos de varias tomas.
13. No deje que caigan objetos o líquidos dentro del aparato – ya que podrían tocar puntos en los que existen niveles de voltaje peligrosos, o causar cortocircuitos que podrían causar incendios o descargas eléctricas.
14. Vea la parte posterior de la caja para indicaciones de seguridad.
15. No coloque en el aparato ninguna llama viva, como por ejemplo una vela.
16. El producto se debe montar en bastidor y debe instalarse según los códigos de montaje locales.
17. El producto no está pensado para aplicaciones marinas.

Important Safety Instructions

Avertissement (Français)

Afin de réduire le risque d'incendie ou de choc électrique, ne pas exposer le système à la pluie ou à l'humidité.

ATTENTION

RISQUE DE CHOC ELECTRIQUE
NE PAS OUVRIR

ATTENTION : AFIN DE REDUIRE LE RISQUE DE CHOC ELECTRIQUE, NE PAS RETIRER LE CAPOT DE L'APPAREIL (OU SON PANNEAU ARRIÈRE). IL N'EXISTE A L'INTERIEUR DE CET EQUIPEMENT AUCUN ELEMENT OU SOUS-ENSEMBLE POUVANT ETRE DEPANNE PAR L'UTILISATEUR. LA MAINTENANCE DOIT ETRE REALISEE PAR UN PERSONNEL QUALIFIE.

Ces signalisations d'AVERTISSEMENT sont peut-être situées à l'arrière del PANARAY® System Digital Controller

L'éclair terminé par une flèche et contenu dans un triangle équilatéral est prévu pour attirer l'attention de l'utilisateur sur la présence à l'intérieur du produit, de tensions dangereuses appliquées à certains éléments non protégés. Ces tensions sont d'une magnitude suffisamment importante pour constituer un risque de choc électrique ou d'électrocution.

Le point d'exclamation contenu dans un triangle équilatéral est prévu pour avertir l'utilisateur sur la présence d'instructions importantes de fonctionnement ou de maintenance (sur le produit et dans ce manuel).

Consignes de sécurité importantes

1. Lisez et appliquez les instructions données dans ce document.
2. Soyez attentif à tous les avertissements—Ceux figurant sur l'appareil et ceux délivrés dans ce guide.
3. N'utilisez pas cet appareil à proximité d'eau ou d'humidité.
4. Pour le nettoyage, utilisez un chiffon sec.
5. N'obstruez aucun des orifices de ventilation. Choisissez un emplacement conforme aux instructions du fabricant—Pour assurer le bon fonctionnement de l'appareil et le protéger contre tout risque de surchauffe, placez celui-ci à un endroit et dans une position qui ne compromette pas sa ventilation.
6. Ne posez pas l'appareil à proximité de sources de chaleur telles que : radiateurs, bouches de chauffage pulsé, poêle ou autres appareils (amplificateurs y compris) dégageant de la chaleur.
7. Pour votre sécurité, ne débranchez pas la liaison de terre sur la prise secteur. Cette prise comporte deux broches mâles et une broche femelle de mise à la terre. Cette dernière sert à vous protéger. Si la fiche équipant votre appareil ne correspond pas à votre prise secteur murale, faites appel à un électricien qualifié pour qu'il effectue son remplacement.
8. Evitez que le cordon d'alimentation ne soit piétiné ou pincé, en particulier au niveau de la prise, d'un bloc multiprises ou de sa sortie de l'appareil.
9. N'utilisez que les accessoires préconisés par le fabricant.
10. N'utilisez qu'un rack, un meuble, un support ou une table préconisé par le fabricant ou vendu pour être utilisé avec l'appareil. Si l'appareil est placé à l'intérieur d'un rack, déplacez celui-ci avec prudence afin d'éviter qu'il n'occasionne des blessures en se renversant.
11. Faites effectuer tous les travaux d'entretien ou de réparation par un personnel qualifié. Une maintenance est nécessaire dès que l'appareil est endommagé de manière quelconque, notamment lorsque le cordon d'alimentation est en mauvais état, qu'un liquide a été renversé ou qu'un objet est tombé à l'intérieur du produit. Il en va de même si l'équipement a été exposé à la pluie ou à l'humidité, qu'il est tombé ou qu'il ne fonctionne pas normalement—N'essayez pas de réparer cet appareil vous-même. En ouvrant ou en retirant un capot, vous risquez d'entrer en contact avec des tensions dangereuses ou de vous retrouver exposé à d'autres dangers. Appelez la société Bose® afin de connaître la station technique agréée la plus proche de chez vous.
12. Pour éviter tout risque d'incendie ou de choc électrique, ne surchargez pas les prises murales, les prolongateurs ou les blocs multiprises.
13. Evitez que des objets ou des liquides ne pénètrent dans l'appareil. Ils pourraient entrer en contact avec des éléments alimentés par des tensions dangereuses ou créer des courts-circuits pouvant provoquer un incendie ou une électrocution.
14. Reportez-vous aux avertissements relatifs à la sécurité, apposés à l'arrière du châs<None><None>sis.
15. Ne placez aucune flamme nue, telle qu'une bougie, sur l'appareil.
16. L'appareil doit être monté sur rack et installé conformément aux réglements locaux du bâtiment.
17. L'appareil n'est pas destiné à être utilisé sur un bateau.

Avvertenza (Italiano)

Per ridurre il pericolo di incendi o scosse elettriche, non esporre l'apparecchio alla pioggia o all'umidità.

ATTENZIONE

PERICOLO DI SCOSSE ELETTRICHE
NON APRIRE

ATTENZIONE: PER RIDURRE IL PERICOLO DI SCOSSE ELETTRICHE, NON RIMUOVERE IL COPERCHIO (O IL PANNELLO POSTERIORE). NON SONO PRESENTI PARTI SOSTITUIBILI DALL'UTENTE. PER L'ASSISTENZA CONTATTATE IL PERSONALE QUALIFICATO.

I seguenti simboli di AVVERTENZA sono apposti sul pannello posteriore del PANARAY® System Digital Controller.

Il simbolo del fulmine all'interno di un triangolo equilatero ha lo scopo di avvertire gli utenti della presenza di tensione pericolosa non isolata all'interno dell'apparecchio, il cui livello è tale da costituire un potenziale rischio di scossa elettrica.

Il simbolo del punto esclamativo all'interno del triangolo equilatero, riportato sull'apparecchio, ha lo scopo di informare gli utenti della presenza di importanti istruzioni di funzionamento e manutenzione nella documentazione fornita con il prodotto.

Importanti norme di sicurezza

1. Leggete attentamente le istruzioni riportate di seguito.
2. Prestate attenzione alle avvertenze riportate sull'apparecchio e nel manuale di istruzioni.
3. Non utilizzate questo apparecchio vicino a fonti di umidità o acqua.
4. Utilizzate esclusivamente un panno asciutto per pulire l'apparecchio.
5. Non ostruite le aperture di ventilazione. Installate l'apparecchio conformemente a quanto indicato nelle istruzioni fornite dal produttore. Al fine di garantire il corretto funzionamento del prodotto e di proteggerlo da un riscaldamento eccessivo, collocate l'apparecchio in una posizione tale da non ostacolarne la corretta aerazione.
6. Non installate l'apparecchio vicino a fonti di calore, quali radiatori, stufe o altri apparati (compresi gli amplificatori) che generano calore.
7. Non manomettete la spina dotata di massa (Shuko), progettata per la vostra sicurezza. Una spina Shuko è composta da due terminali e da un terzo terminale a forca per la massa aggiunto per garantire un più elevato livello di sicurezza. Se la spina fornita in dotazione non è adatta alle vostre prese a muro, contattare un tecnico per sostituire le prese.
8. Collocate il cavo di alimentazione in modo da evitare che venga calpestato o piega to, in particolare in corrispondenza delle spine, delle prese multiple e dei punti dove questo esce dall'apparecchio.
9. Utilizzate esclusivamente gli accessori specificati dal produttore.
10. Utilizzate esclusivamente con supporti mobili, piedistalli, staffe o ripiani specificati dal produttore o forniti in dotazione con l'apparecchio. Quando utilizzate un supporto mobile, prestate attenzione durante gli spostamenti dell'insieme carrello/apparecchio per evitare danni fisici causati da un possibile ribaltamento.
11. Per le riparazioni rivolgetevi al personale di assistenza qualificato. Contattate il servizio di assistenza nel caso l'apparecchio presenti danni di qualsiasi tipo, ad esempio al cavo di alimentazione o alla spina oppure dovuti alla penetrazione di liq uido o di altro materiale all'interno dell'apparecchio, all'esposizione dell'apparecchio alla pioggia o all'umidità, a malfunzionamenti o, infine, a caduta. Non tentate di riparare da soli il prodotto. L'apertura o la rimozione del coperchio potrebbe esporre l'utente a tensioni pericolose o ad ulteriori rischi. Vi suggeriamo di contattare la Bose® per informazioni relative al centro di assistenza autorizzato più vicino.
12. Per prevenire il pericolo di incendi o scosse elettriche, evitate il sovraccarico delle prese a muro, delle prolunghe o delle prese multiple di corrente.
13. Prestate attenzione che del liquido o altro materiale non penetri all'interno del prodotto poiché potrebbe venire a contatto con parti sotto tensione o soggette a cortocircuito con conseguente possibilità di incendi o di scosse elettriche.
14. I marchi delle certificazioni di sicurezza sono apposti sul pannello posteriore del prodotto.
15. Non avvicinate all'apparecchio alcun tipo di fiamma, ad esempio un candeliere acceso.
16. Il prodotto deve essere montato in rack e installato in linea con le disposizioni vigen ti in materia edilizia.
17. Il prodotto non è adatto per l'utilizzo in ambito marittimo.

Important Safety Instructions

1. Read these instructions.
2. Keep these instructions – for future reference.
3. Heed all warnings – on the product and in the owner's guide.
4. Follow all instructions.
5. Do not use this apparatus near water or moisture.
6. Clean only with a dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions – To ensure reliable operation of the product and to protect it from overheating, put the product in a position and location that will not interfere with its proper ventilation.
8. Do not install near any heat sources, such as radiators, heat registers, stoves or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding-type plug has two blades and a third grounding prong. The wider blade or third prong are provided for your safety. If the provided plug does not fit in your outlet, consult an electrician for replacement of the obsolete outlet.
10. Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Only use attachments/accessories specified by the manufacturer.
12. Use only with the cart, stand, tripod, bracket or table specified by the manufacturer or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.

13. Unplug this apparatus during lightning storms or when unused for long periods of time – to prevent damage to this product.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way: such as power-supply cord or plug is damaged; liquid has been spilled or objects have fallen into the apparatus; the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped – Do not attempt to service this product yourself. Opening or removing covers may expose you to dangerous voltages or other hazards. Please call Bose® to be referred to an authorized service center near you.
15. To prevent risk of fire or electric shock, avoid over loading wall outlets, extension cords, or integral convenience receptacles.
16. Do not let objects or liquids enter the product – as they may touch dangerous voltage points or short-out parts that could result in a fire or electric shock.
17. See product enclosure back for safety related markings.
18. No naked flame sources, such as lighted candles, should be placed on the apparatus.

Information about products that generate electrical noise

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, this is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, you are encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a different circuit than the one to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Note: *Unauthorized modification of the receiver or radio remote control could void the user's authority to operate this equipment.*

This product complies with the Canadian ICES-003 Class A specifications.

1.0 Introduction

Thank you for purchasing the Bose® PANARAY® system digital controller.

The PANARAY system digital controller uses a digital signal processing (DSP) architecture to provide smooth, accurate spectral response for Bose professional loudspeakers that require active equalization.

Bose professional loudspeaker EQ can be applied by simply selecting the appropriate preset and configuration. No additional programming is required.

The PANARAY system digital controller has two balanced XLR analog inputs and four balanced XLR analog outputs. It can be configured to run two mono Bose speaker EQ presets or a single stereo Bose speaker EQ preset (including preset crossover settings).

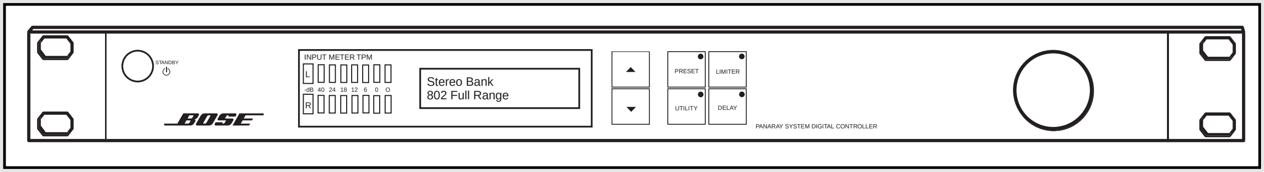
There are a total of 180 presets, which are separated into Stereo, Mono, and Bass Array Banks. All routing is automatic, based on the selected preset and configuration. See the preset chart in section 13.0 for a full list of presets with descriptions and output routing defaults.

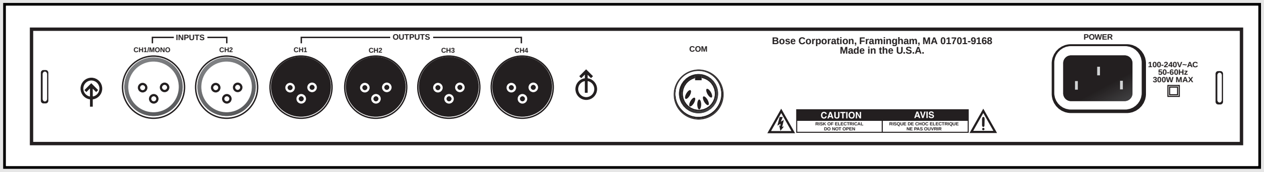
In addition to providing Bose speaker equalization, the PANARAY system digital controller also has the following easy-to-use programmable features:

- Four independent programmable line delays (200 ms per output) that can be used for decentralized or distributed speaker alignment and for creating custom bass arrays.

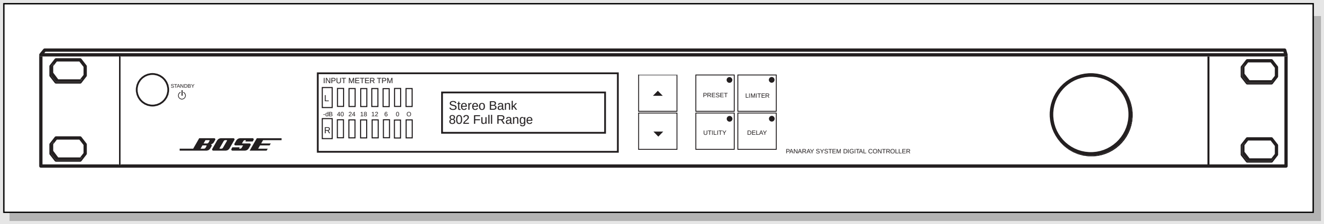
- Four independent programmable output limiters that can be set to protect speakers from unpredictable spikes in program material.

- Programmable signal router that allows each output to select Full Range, High Pass, Low Pass or Bypass EQ combinations of input signals.


STANDBY
BOSE
INPUT METER TPM
L
-dB 40 24 18 12 6 0 O
R
Stereo Bank
802 Full Range
PRESET
LIMITER
UTILITY
DELAY
PANARAY SYSTEM DIGITAL CONTROLLER


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
CAUTION
RISK OF ELECTRICAL
DO NOT OPEN
AVIS
RISQUE DE CHOC ELECTRIQUE
NE PAS OUVRIR
POWER
100-240V~AC
50-60Hz
300W MAX

2.0 Front Panel


STANDBY
INPUT METER TPM
L
-dB 40 24 18 12 6 0 O
R
Stereo Bank
802 Full Range
PRESET
LIMITER
UTILITY
DELAY
BOSE
PANARAY SYSTEM DIGITAL CONTROLLER

STANDBY button:
Standby on/off

Input Meters:
The peak meter shows the input level of left and right channels. The meter range is:
O (Overload), 0, -6, -12, -24, -40dB

Overload LEDs:
The Overload LEDs indicate one of two situations.
- The input level is clipping (you will hear it).
- There is an internal DSP overflow.

The Overload LED is lit when one sample is at 0 dBFS.

Display:
The 2 x 16 LCD displays Preset names, parameter names, and values depending on the mode of operation.

Arrow UP Button:
Press to navigate up through the user interface.

Arrow DOWN Button:
Press to navigate down through the user interface.

PRESET Button:
Press to enter preset mode, which provides access to Bose® speaker EQ presets.

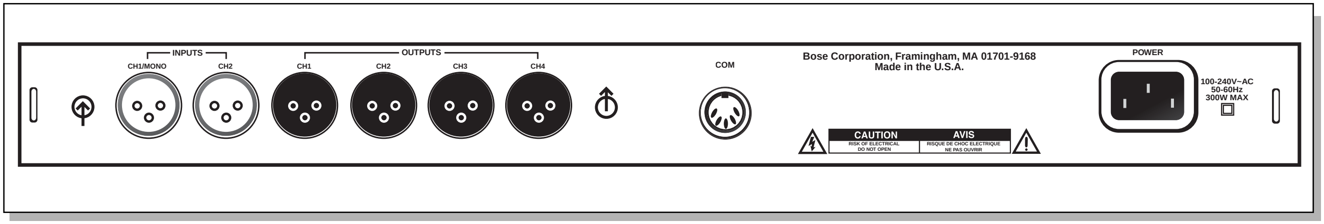
LIMITER Button:
Press to access output limiter parameters.

UTILITY Button:
Press to access ROUTING, LEVEL and general SYSTEM parameters.

DELAY Button:
Press to access output delay parameters.

Encoder Wheel:
Selects presets and changes parameter values.

3.0 Rear Panel


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
CAUTION
RISK OF ELECTRICAL
DO NOT OPEN
AVIS
RISQUE DE CHOC ELECTRIQUE
NE PAS OUVRIR
POWER
100-240V~AC
50-60Hz
300W MAX

Balanced XLR Inputs:
Two balanced XLR inputs

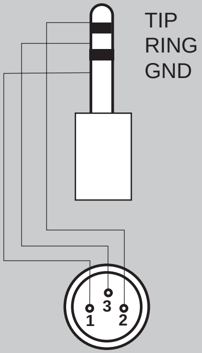
Balanced XLR Outputs:
Four balanced XLR outputs

COM Port:
5-pin DIN that will be used for future software upgrades or for adding new Bose® speaker EQ presets

Power Input:
100-240VAC switching power supply input

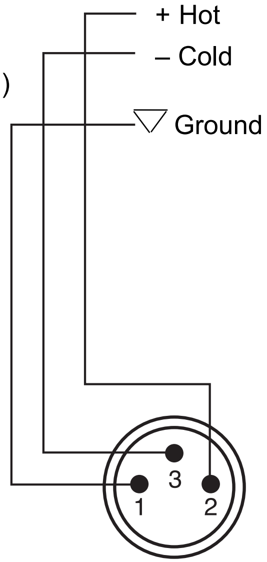
Jack (balanced) - XLR

Sleeve - Pin 1 (Ground)
Tip - Pin 2 (Hot)
Ring - Pin 3 (Cold)


TIP
RING
GND
3
1
2

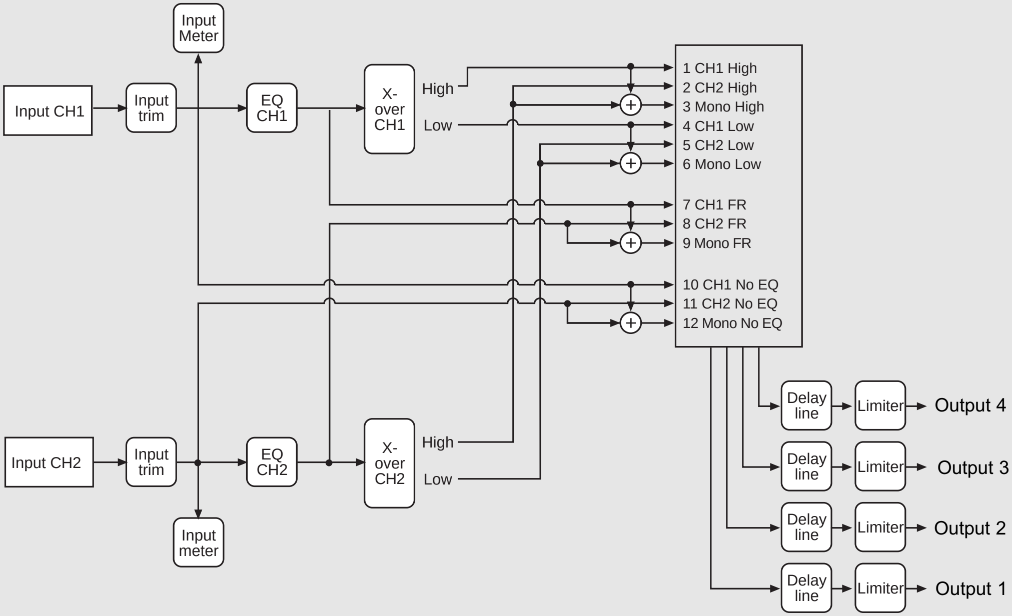
XLR Wiring
Jack (balanced) - XLR

Sleeve - Pin 1 (Ground)
Tip - Pin 2 (Hot)
Ring - Pin 3 (Cold)


+ Hot
– Cold
Ground
3
1
2

4.0 Signal Flow

The PANARAY® system digital controller has the ability to run a single stereo preset, a single mono preset, or to run in dual mono mode. In the following section, the controller's signal flow is broken down to provide a better understanding of its full functionality.



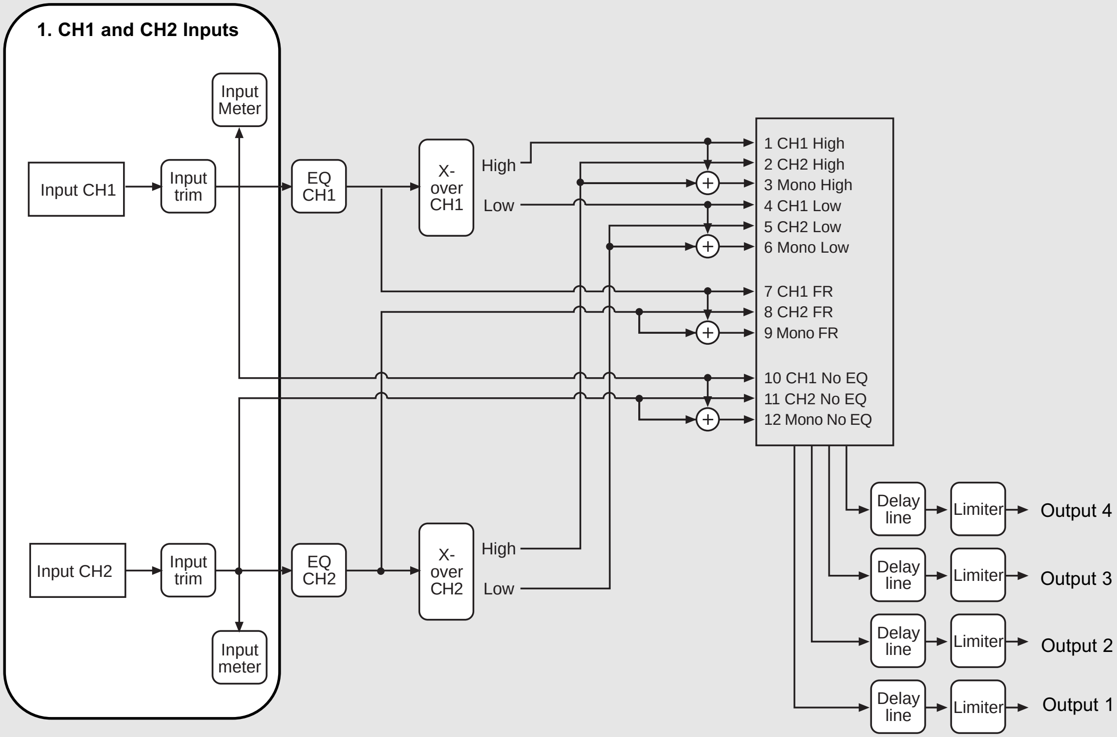
Input Meter
Input CH1
Input trim
EQ CH1
X-over CH1
High
Low
1 CH1 High
2 CH2 High
3 Mono High
4 CH1 Low
5 CH2 Low
6 Mono Low
7 CH1 FR
8 CH2 FR
9 Mono FR
10 CH1 No EQ
11 CH2 No EQ
12 Mono No EQ
Delay line
Limiter
Output 4
Output 3
Output 2
Output 1
Input CH2
Input trim
EQ CH2
X-over CH2
High
Low
Input meter

Stereo Operation: When a preset is loaded from the stereo bank, the EQ/crossover blocks both process each input the same way, meaning that the parameters are "linked". The signal then passes through the router section where the CH1 and CH2 inputs are sent to independent output channels for stereo operation.

Mono Operation: When a preset is loaded from the mono bank, the EQ/crossover blocks both process each input the same way, meaning that the parameters are "linked". The signal then passes through the router section where the CH1 and CH2 inputs are summed to mono. The controller has the ability to sum a stereo signal to mono, or, it can process a single input for mono operation. Audio information is routed to the same outputs regardless of the number of inputs that are connected.

Dual Mono Operation: When the PANARAY® system digital controller is used in dual mono mode, the EQ/crossover blocks process each input independently – allowing the user to run two independent speaker EQs. The router also processes each input independently, so each output is discrete.

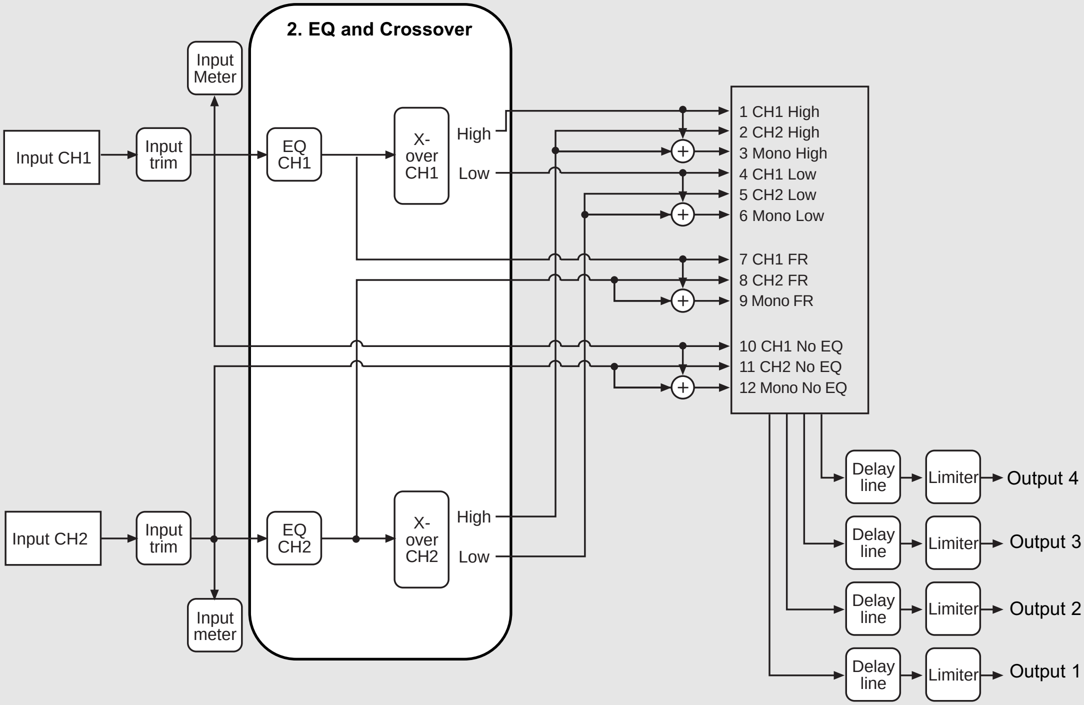
4.1 Signal Flow – CH1 and CH2 Inputs


1. CH1 and CH2 Inputs
Input Meter
Input CH1
Input trim
EQ CH1
X-over CH1
High
Low
Input CH2
Input trim
EQ CH2
X-over CH2
High
Low
Input meter
1 CH1 High
2 CH2 High
3 Mono High
4 CH1 Low
5 CH2 Low
6 Mono Low
7 CH1 FR
8 CH2 FR
9 Mono FR
10 CH1 No EQ
11 CH2 No EQ
12 Mono No EQ
Delay line
Limiter
Output 4
Delay line
Limiter
Output 3
Delay line
Limiter
Output 2
Delay line
Limiter
Output 1

1. CH1 and CH2 Input Section:

A stereo, mono, or dual mono signal passes through the input section.

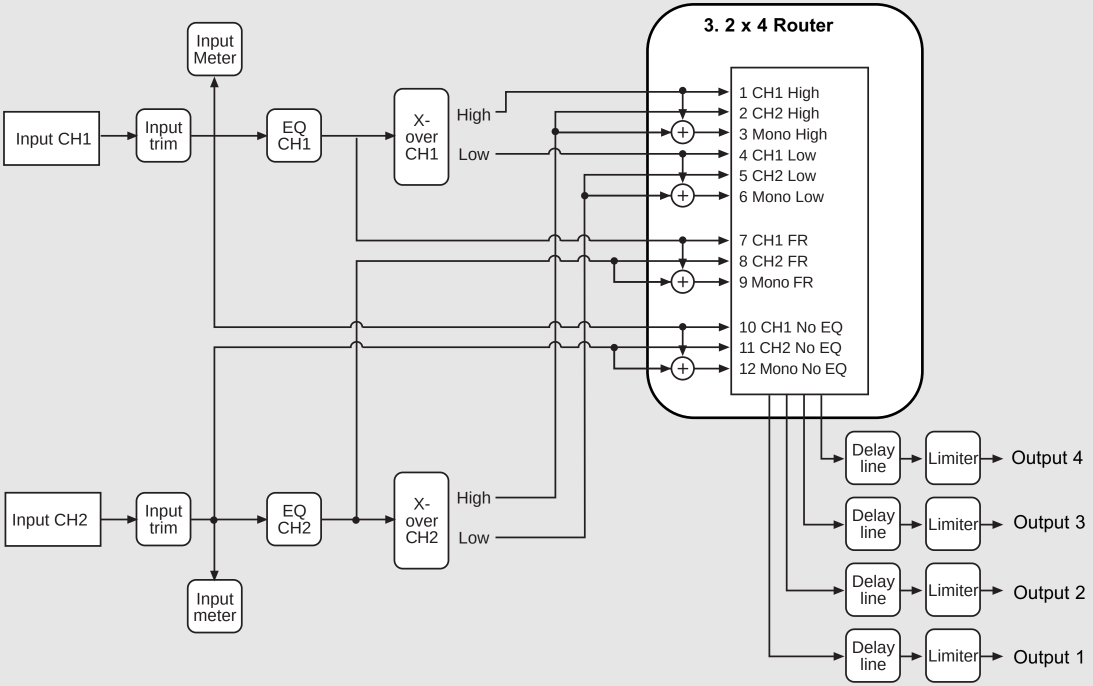
4.2 Signal Flow – EQ and Crossover


2. EQ and Crossover
Input Meter
Input CH1
Input trim
EQ CH1
X-over CH1
High
Low
Input CH2
Input trim
EQ CH2
X-over CH2
High
Low
Input meter
1 CH1 High
2 CH2 High
3 Mono High
4 CH1 Low
5 CH2 Low
6 Mono Low
7 CH1 FR
8 CH2 FR
9 Mono FR
10 CH1 No EQ
11 CH2 No EQ
12 Mono No EQ
Delay line
Limiter
Output 4
Delay line
Limiter
Output 3
Delay line
Limiter
Output 2
Delay line
Limiter
Output 1

2. EQ and Crossover:

Next, the program material passes through the EQ and crossover section.

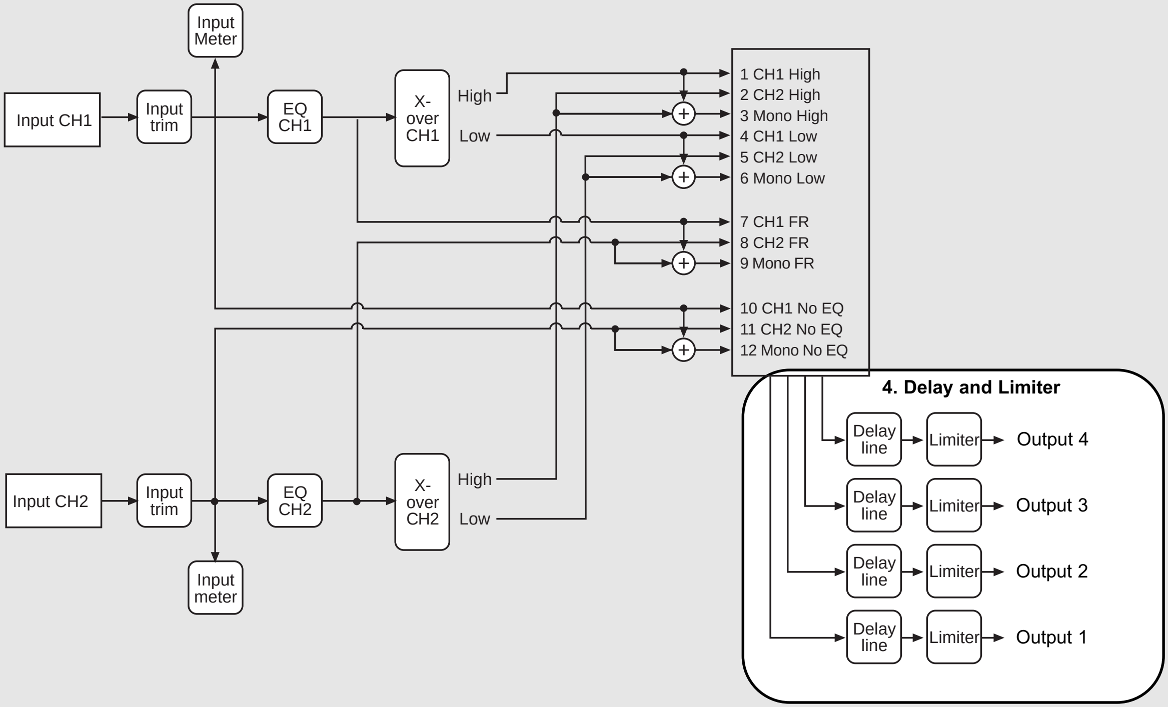
4.3 Signal Flow – 2 x 4 Router


3. 2 x 4 Router
Input Meter
Input CH1
Input trim
EQ CH1
X-over CH1
High
Low
1 CH1 High
2 CH2 High
3 Mono High
4 CH1 Low
5 CH2 Low
6 Mono Low
7 CH1 FR
8 CH2 FR
9 Mono FR
10 CH1 No EQ
11 CH2 No EQ
12 Mono No EQ
Input CH2
Input trim
EQ CH2
X-over CH2
High
Low
Input meter
Delay line
Limiter
Output 4
Delay line
Limiter
Output 3
Delay line
Limiter
Output 2
Delay line
Limiter
Output 1

3. 2 x 4 Router:

The 2 x 4 router gives the user the ability to select from the options listed in the routing matrix to the right, and route the selection to any of the controller’s four outputs. Complete detail of the router operation can be found in section 9.0.

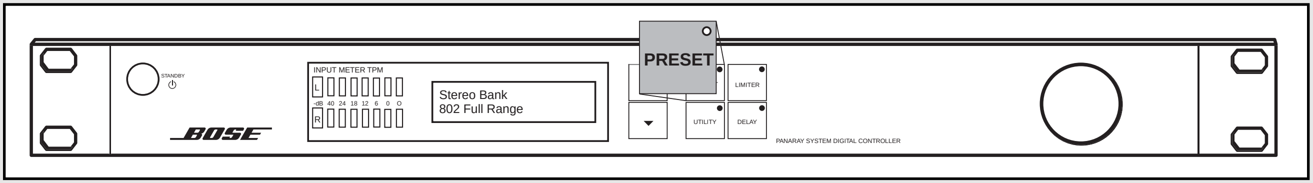
4.4 Signal Flow – Delay and Limiter


Input Meter
Input CH1
Input trim
EQ CH1
X-over CH1
High
Low
1 CH1 High
2 CH2 High
3 Mono High
4 CH1 Low
5 CH2 Low
6 Mono Low
7 CH1 FR
8 CH2 FR
9 Mono FR
10 CH1 No EQ
11 CH2 No EQ
12 Mono No EQ
4. Delay and Limiter
Delay line
Limiter
Output 4
Output 3
Output 2
Output 1
Input CH2
Input trim
EQ CH2
X-over CH2
High
Low
Input meter

4. Delay and Limiter:

After the signal passes through the routing matrix, it goes to each of the four outputs and passes through the delay and limiter blocks on each channel. The delay and limiter blocks can be bypassed by the user. For more information on controlling the delay and limiter see sections 6.0 and 7.0.

5.0 PRESET BUTTON


PRESET
STANDBY
INPUT METER TPM
L
R
-dB 40 24 18 12 6 0 O
Stereo Bank
802 Full Range
LIMITER
UTILITY
DELAY
BOSE
PANARAY SYSTEM DIGITAL CONTROLLER

There are 180 speaker EQ presets in the PANARAY® system digital controller. They are organized into banks: Stereo, Mono, Bass Array and Dual Mono. Within each bank are presets that provide Bose® speaker equalization.

To load a preset from the front panel:

- Press the PRESET button.
  Next, use the Arrow UP or Arrow DOWN button to select the configuration bank (Stereo, Mono, etc.).

- Then use the Encoder to select the desired preset. The PANARAY system digital controller automatically updates the outputs to reflect the selected EQ preset and configuration.

The examples on the following pages illustrate how the PANARAY system digital controller is configured for each type of default output. A complete list of presets (with descriptions) can be found in section 13.0.

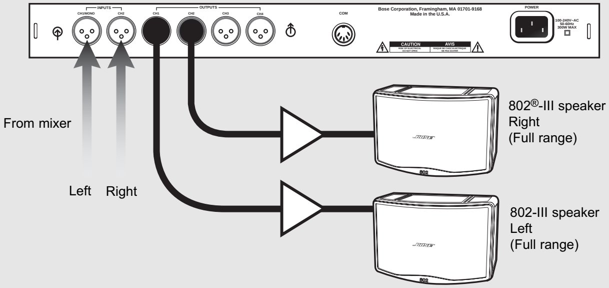
5.1 PRESET Button – Stereo Bank

Stereo Preset Bank: Presets designed for Stereo operation.

Example 1: Stereo Full Range

PANARAY® System Digital Controller Display

```
Stereo
802III
```


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
POWER
100-240V~AC 50-60Hz 200W MAX
CAUTION
AVIS
From mixer
Left
Right
802®-III speaker
Right
(Full range)
802-III speaker
Left
(Full range)

Preset Description

802®-III Speaker Stereo Full Range

Controller Inputs

Ch 1/Mono	Left from mixer
Ch 2	Right from mixer

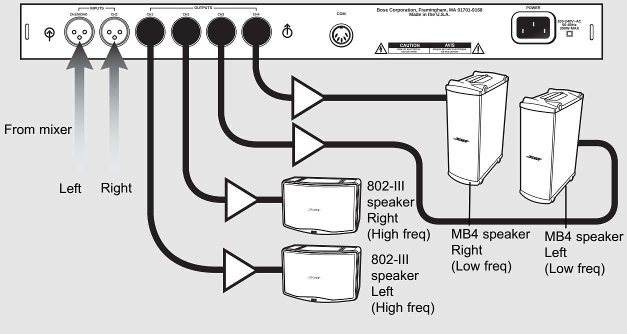
Controller Default Output Routing

	Ch 1	Ch 2	Ch 3	Ch 4
Routing	802-III speaker CH1 (Full range)	802-III speaker CH2 (Full range)	802-III speaker CH1 (Full range)	802-III speaker CH2 (Full range)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off

Example 2: Stereo Mid-High with Stereo Bass

PANARAY System Digital Controller Display

```
Stereo
802III+MB4
```


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
POWER
100-240V~AC 50-60Hz 300W MAX
CAUTION
AVIS
From mixer
Left
Right
802-III
speaker
Right
(High freq)
802-III
speaker
Left
(High freq)
MB4 speaker
Right
(Low freq)
MB4 speaker
Left
(Low freq)

Preset Description

802-III Speaker Stereo with MB4 Speaker Stereo Bass

Controller Inputs

Ch 1/Mono	Left from mixer
Ch 2	Right from mixer

Controller Default Output Routing

	Ch 1	Ch 2	Ch3	Ch4
Routing	802-III speaker CH1 (High freq)	802-III speaker CH2 (High freq)	MB4 speaker CH1 (Low freq)	MB4 speaker CH2 (Low freq)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off

5.1.1 PRESET Button – Stereo Bank

Example 3: Stereo Mid-High with Mono Bass

Due to space limitations in the PANARAY® system digital controller, there are no Presets that default to a Stereo Mid-High with Mono Bass configuration. To set up the controller for this operation, simply perform the following steps.

1. Select the desired Preset from the Stereo bank.

2. Press the UTILITY Button and use the Arrow DOWN button to navigate to Output Route 3 and Output Route 4. Set both outputs to Mono/Low.

Now all of the bass content is set up to be summed to mono, but the mid-high devices will still be stereo.
Note: Save all changes before loading a new Preset. Unsaved router changes will be lost if another speaker EQ Preset is loaded. To save settings see section 11.0.

PANARAY System Digital Controller Display

```
Stereo
402II+MB4
```

Preset Description

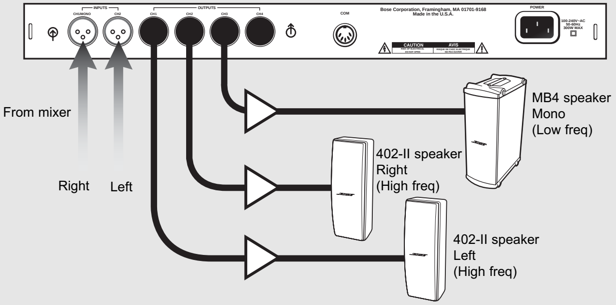
402□-II Speaker Stereo with MB4 Speaker Mono

Controller Inputs

Ch 1/Mono	Left from mixer
Ch 2	Right from mixer

Controller Default Output Routing

	Ch 1	Ch 2	Ch3	Ch4
Routing	402-II speaker CH1 (High freq)	402-II speaker CH2 (High freq)	MB4 speaker CH1 (Low freq)	MB4 speaker CH2 (Low freq)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off


INPUTS
OUTPUTS
CH1/MONO
CH2
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
POWER
CAUTION
AVIS
From mixer
Right
Left
MB4 speaker
Mono
(Low freq)
402-II speaker
Right
(High freq)
402-II speaker
Left
(High freq)

5.2 PRESET Button – Mono Bank

Mono Preset Bank: Presets designed for Mono Operation.

Example 1: Mono Full Range

PANARAY® System Digital Controller Display

```
Mono
402II
```

Preset Description

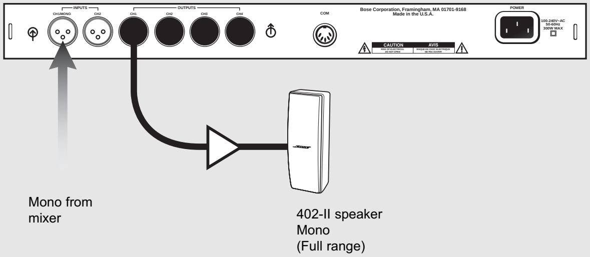
402□ -II Speaker Mono Full Range

Controller Inputs

Ch 1/Mono	Mono from mixer
Ch 2	No input

Controller Default Output Routing

	Ch 1	Ch 2	Ch3	Ch4
Routing	402-II speaker Mono (Full range)	402-II speaker Mono (Full range)	402-II speaker Mono (Full range)	402-II speaker Mono (Full range)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
CAUTION
AVIS
POWER
100-240V~AC
50-60Hz
300W MAX
Mono from mixer
402-II speaker
Mono
(Full range)

Example 2: Mono Mid-High Device with Mono Bass

PANARAY System Digital Controller Display

```
Mono
502A+502B
```

Preset Description

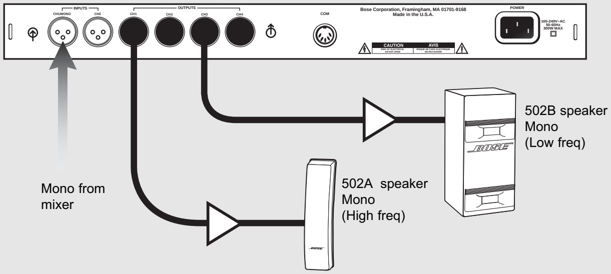
502® A Speaker Mono with 502B Speaker Mono

Controller Inputs

Ch 1/Mono	Mono from mixer
Ch 2	No input

Controller Default Output Routing

	Ch 1	Ch 2	Ch3	Ch4
Routing	502A speaker Mono (High freq)	502A speaker Mono (High freq)	502B speaker Mono (Low freq)	502B speaker Mono (Low freq)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
CAUTION
AVIS
POWER
100-240V~AC
50-60Hz
300W MAX
502B speaker
Mono
(Low freq)
Mono from mixer
502A speaker
Mono
(High freq)

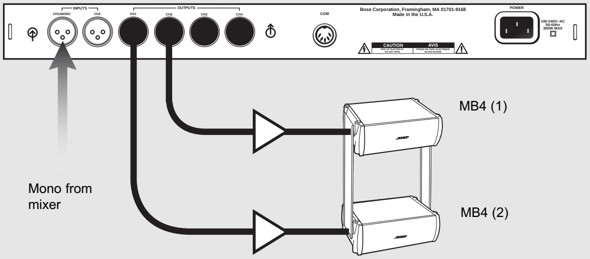
5.3 PRESET Button – Bass Array Bank

Bass Array Presets:

Example 1: 2x Broadside Bass Array

PANARAY® System Digital Controller Display

```
Bass Array
MB4 2XBS180Hz
```


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
CAUTION
RISK OF ELECTRICAL
DO NOT OPEN
AVIS
RISQUE DE CHOC ELECTRIQUE
NE PAS OUVRIR
POWER
100-240V~AC
50-60Hz
300W MAX
MB4 (1)
Mono from mixer
MB4 (2)

Preset Description

Preset: MB4 2x Broadside Bass Array

Controller Inputs

Ch 1	Mono source

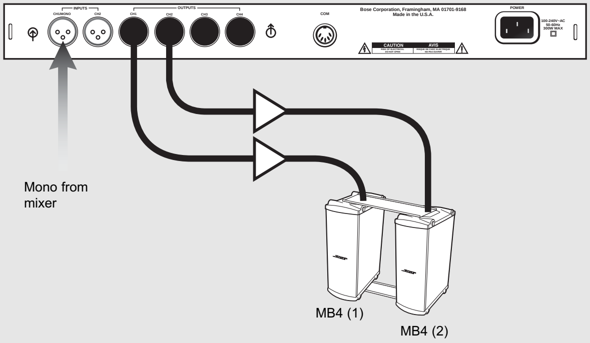
Controller Default Output Routing

	Ch 1	Ch 2	Ch 3	Ch 4
Routing	MB4 (1) Mono	MB4 (2) Mono	N/A	N/A
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off

Example 2: 2x Endfire Bass Array

PANARAY System Digital Controller Display

```
Bass Array
MB4 2XEF180Hz
```


INPUTS
CH1/MONO
CH2
OUTPUTS
CH1
CH2
CH3
CH4
COM
Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
CAUTION
AVIS
POWER
100-240V~AC
50-60Hz
300W MAX
Mono from mixer
MB4 (1)
MB4 (2)

Preset Description

Preset: MB4 2x Endfire Bass Array

Controller Inputs

Ch 1	Mono source

Controller Default Output Routing

	Ch 1	Ch 2	Ch 3	Ch 4
Routing	MB4 (1) Mono	MB4 (2) Mono	N/A	N/A
Limiter	Off	Off	Off	Off
Delay	On	On	Off	Off

5.3 PRESET Button – Bass Array Bank

Example 2: 4x Endfire Bass Array

PANARAY® System Digital Controller Display

```
Bass Array
MB4 4XEF180Hz
```

Preset Description

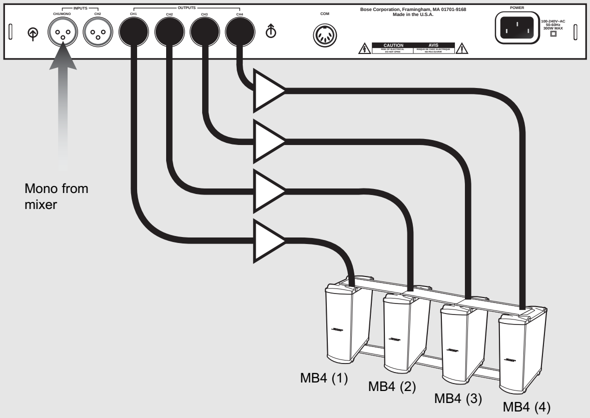
Preset: MB4 4x Endfire Bass Array

Controller Inputs

Ch 1	Mono source

Controller Default Output Routing

	Ch 1	Ch 2	Ch 3	Ch 4
Routing	MB4 (1) mono	MB4 (2) mono	MB4 (3) mono	MB4 (4) mono
Limiter	Off	Off	Off	Off
Delay	On	On	On	On


Bose Corporation, Framingham, MA 01701-9168
Made in the U.S.A.
INPUTS
OUTPUTS
CH1
CH2
CH3
CH4
COM
POWER
CAUTION
AVIS
Mono from mixer
MB4 (1)
MB4 (2)
MB4 (3)
MB4 (4)

5.4 PRESET Button – Dual Mono Operation

Loading presets from Dual Mono mode:

Dual Mono: When the PANARAY® system digital controller is used in Dual Mono mode, the EQ/Crossover blocks process each input independently, allowing you to run two different speaker EQs.

To access Dual Mono mode, press the PRESET button. Then press the Arrow DOWN button until you see the following display:

```
1: .402I
2:  402I
```

The period (.) next to the preset name indicates that the Encoder can change the preset. To select between 1 and 2, use the Arrow UP/Arrow Down buttons.

CH1 Input
The top of the display shows the preset name that will be processed by the CH1 input. This signal will be routed to Outputs CH1 and CH2. If this is a full-range preset, the same EQ will be applied to Outputs CH1 and CH2. If this preset consists of a mid-high device with a bass loudspeaker, the mid-high frequencies will be routed to Output CH1 and the low frequencies will be routed to Output CH2.

CH2 Input
The bottom of the display shows the preset that is processed by the CH2 input. This signal will be routed to Outputs CH3 and CH4. If this is a full-range preset, the same EQ will be applied to Outputs CH3 and CH4. If this preset consists of a mid-high device with a bass loudspeaker, the mid-high frequencies will be routed to Output CH3 and the low frequencies will be routed to Output CH4.

Note: If you only have a connection plugged into Input 1, the signal will only pass through Outputs 1 and 2.

If you only have a signal plugged into Input 2, the signal will only pass through Outputs 3 and 4.

If you want to send the same audio information to both of the PANARAY system digital controller's inputs, you can achieve this in two ways:

1. From your mixer set all of your pan controls to center position (this will sum your information to mono). Next plug the left and right outputs from your mixer into CH1 and CH2 on the PANARAY system digital controller.

2. From a mono source use a Y-cable to send the same information into CH1 and Ch2 on the PANARAY system digital controller.

5.4 PRESET Button – Dual Mono Bank

Dual Mono Presets: Independent selection of two single mono presets.

Example 1: Dual Mono

PANARAY® System Digital Controller Display

```
1: 802III+502B
2: 402II+MB4
```

Presets

Preset 1: 802□-III Speaker with 502B Speaker Mono

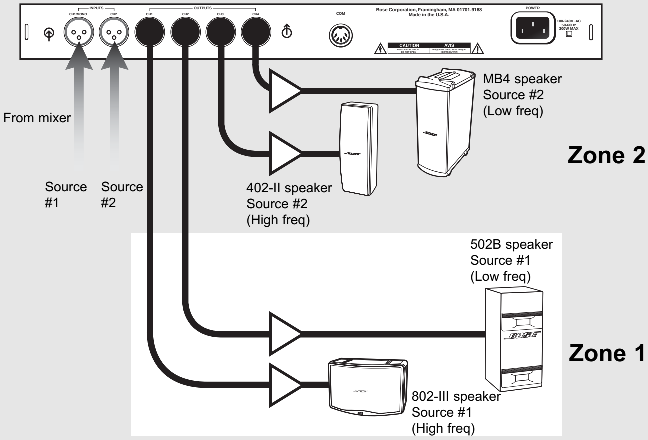
Preset 2: 402□-II Speaker with MB4 Speaker Mono

Controller Inputs

Ch 1	Source #1 from mixer
Ch 2	Source #2 from mixer

Controller Default Output Routing

	Ch 1	Ch 2	Ch3	Ch4
Routing	802-III speaker Source #1 (High freq)	502B speaker Source #1 (Low freq)	402-II speaker Source #2 (High freq)	MB4 speaker Source #2 (Low freq)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off


From mixer
Source #1
Source #2
MB4 speaker
Source #2
(Low freq)
402-II speaker
Source #2
(High freq)
Zone 2
502B speaker
Source #1
(Low freq)
802-III speaker
Source #1
(High freq)
Zone 1

Example 2: Dual Mono

PANARAY System Digital Controller Display

```
1: 502A+502B
2: 402II
```

Preset Description

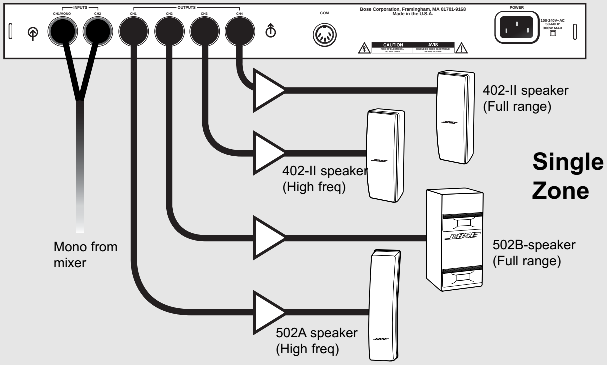
Preset 1: 502□A Speaker with 502B Speaker Mono
Preset 2: 402□-II

Controller Inputs

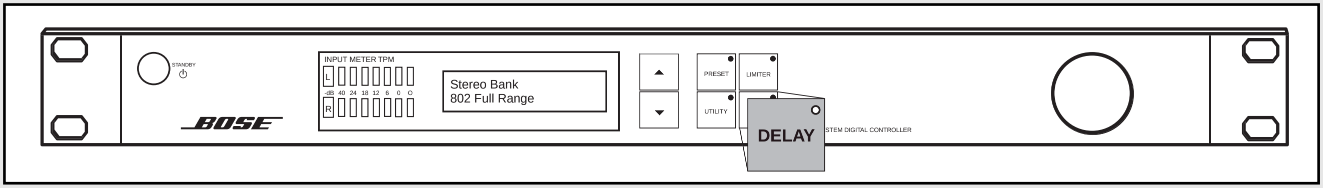
Ch 1	Mono source (from Y-cable)
Ch 2	Mono source (from Y-cable)

Controller Default Output Routing

	Ch 1	Ch 2	Ch3	Ch4
Routing	502A (High freq)	502B (Low freq)	402II (Full range)	402II (Full range)
Limiter	Off	Off	Off	Off
Delay	Off	Off	Off	Off


Mono from mixer
402-II speaker
(Full range)
402-II speaker
(High freq)
Single Zone
502B-speaker
(Full range)
502A speaker
(High freq)

6.0 DELAY Button


STANDBY
BOSE
INPUT METER TPM
L
R
-dB 40 24 18 12 6 0 O
Stereo Bank
802 Full Range
PRESET
LIMITER
UTILITY
DELAY
STEM DIGITAL CONTROLLER

Pressing the DELAY button provides access to four independent output delays. There is a single delay line available on each of the controller's four outputs. The delay function is always available regardless of the mode of operation.

Each delay provides up to 200ms of delay on each of the four outputs (increments in 0.02ms intervals).

Use the Arrow UP and Arrow DOWN buttons to navigate through the parameters, and use the Encoder wheel to select parameter values.

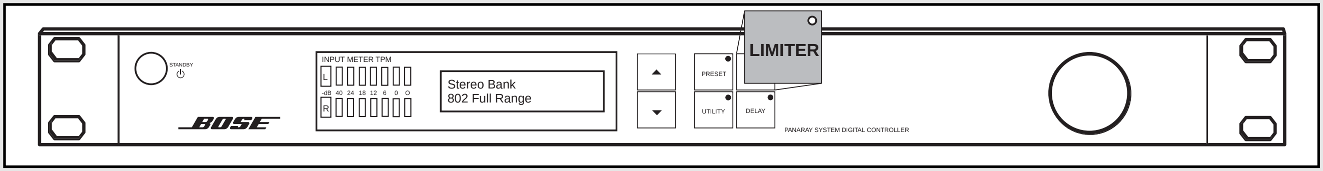
The four output delays have the following parameters:

Delay On/Off: Bypasses delay

Delay Time: Adjustable in 0.02ms increments

Delay Parameter	Range
Delay 1	On/Off
Delay 2	On/Off
Delay 3	On/Off
Delay 4	On/Off
Delay time 1	0.00ms - 200.00ms
Delay time 2	0.00ms - 200.00ms
Delay time 3	0.00ms - 200.00ms
Delay time 4	0.00ms - 200.00ms

7.0 LIMITER Button



Pressing the LIMITER button provides access to four independent output limiters that can protect the loudspeakers from damage if an operator overdrives the system. Use the Arrow UP and Arrow DOWN buttons to navigate through the parameters, and use the ENCODER wheel to select parameter values.

The four output limiters each have the the following parameters:

Limiter On/Off: Bypasses the limiter

Threshold: Sets the level at which the limiter will activate. When the input signal exceeds the threshold, the limiter will be activated. The lower the threshold, the more sensitive the limiting will be.

Ratio: The ratio of gain reduction.
For example: if the ratio is set to 4:1, it means that for every 4dB the actual audio signal will rise only 1dB over the threshold.

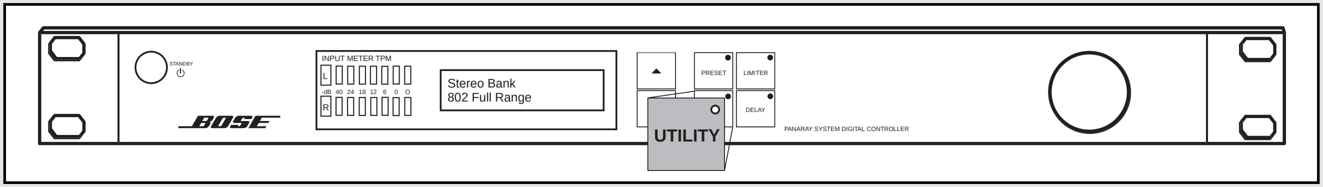
Attack: The time it takes the limiter to reach gain reduction specified by the ratio parameter when the signal is above the set threshold.

Release: Release is the time that the limiter uses to release the gain reduction when the signal exceeds the threshold.

Gain: Use the gain parameter to compensate for unwanted gain reduction caused by heavy limiting.

Limiter Parameter	Range
Lim1	On/Off
Lim2	On/Off
Lim3	On/Off
Lim4	On/Off
Lim1 Threshold	off - inf:1
Lim1 Ratio	off - inf:1
Lim1 Attack	1.0ms - 100ms
Lim1 Release	100ms - 7.0s
Lim1 Gain	-100dB - 18dB
Lim2 Threshold	-40bB - 0dB
Lim2 Ratio	off - inf:1
Lim2 Attack	1.0ms - 100ms
Lim2 Release	100ms - 7.0s
Lim2 Gain	-100dB - 18dB
Lim3 Threshold	-40bB - 0dB
Lim3 Ratio	off - inf:1
Lim3 Attack	1.0ms - 100ms
Lim3 Release	100ms - 7.0s
Lim3 Gain	-100dB - 18dB
Lim4 Threshold	-40bB - 0dB
Lim4 Ratio	off - inf:1
Lim4 Attack	1.0ms - 100ms
Lim4 Release	100ms - 7.0s
Lim4 Gain	-100dB - 18dB

8.0 UTILITY Button



Pressing the UTILITY button provides access to the Level, System and Routing parameters. Use the Arrow UP/Arrow DOWN buttons to navigate through the parameters, and use the Encoder wheel to select parameter values.

Parameter Definitions:

Output Level

Controls the overall output level of outputs 1, 2, 3, and 4.

Parameter Range: Off; -97dB to 0dB

Default Value: 0dB

Output Range

You can use this parameter to match the input level of other equipment.

Parameter Range: 2dB, 8dB (consumer), 14dB, and 20dB (pro)

Default Value: 20dB

Allows you to adjust the maximum output level of the PANARAY® system digital controller.

Input Sensitivity

Sets level in dBu to allow full-scale operation. A digital product produces the best signal-to-noise performance when operating at 0 dBFS (full scale). For example: For a mixer with +4dBu outputs, the Input Sensitivity should be set to +8dBu to account for peaks in the programming material.

Parameter Range: 0dBu – 24dBu

Default Value: 20dBu

Input Trim CH1

Allows you to change the input level in 1dB steps on CH1.

Parameter Range: -40dB to 0dB

Default Value: 20dB

Input Trim CH2

Allows you to change the input level in 1dB steps on CH2.

Parameter Range: -40dB to 0dB

Default Value: 20dB

Output Route 1
Output Route 2
Output Route 3
Output Route 4

Note: See Section 9.0 on page 29 for details on router functionality.

Delay Unit

Selectable between milliseconds, feet, and meters.

Default Value: milliseconds

8.0 UTILITY Button

Delay Master:
Selectable between PRESET and GLOBAL.

When set to PRESET, the delay values will be reset each time a new preset is selected and will match the delay values stored in the original preset.

When set to GLOBAL, the delay values remain fixed atthe current settings even if a new preset is loaded.

Default Value: PRESET

Limiter Master:
Selectable between PRESET or GLOBAL.
When set to PRESET, the limiter values will be reset each time a new preset is selected and will match the limiter values stored in the original preset.

When set to GLOBAL, the limiter values remain fixed at the current settings even if a new preset is loaded.

Default Value: PRESET

Input Configuration:
Selectable between Stereo, CH1+CH2 (sum), or CH1 (mono).

Stereo: Optimizes input gain for stereo operation
CH1+CH2 (sum): Optimizes the input gain when the controller is used to sum CH1 and CH2 to MONO.

CH1 (mono): Optimizes the input gain when a single input is used for MONO operation.

Note: When this option is selected CH2 input does not work.

Default Value: Stereo

Lock Out:
Turns user lockout on and off.
Note: See section 8.1 for more details.

Default Value: Off

Custom Presets:
Provides access to stored presets. Selectable between ON and OFF.

Default Value: Off

View Angle:
Adjusts display contrast. Selectable 0 – 7.

Default Value: 3

8.1 UTILITY Button – User Lockout

User Lockout:

The User Lockout feature is available to prevent others from tampering with the EQs and associated parameters that have been set.

To access the User Lockout feature, press the UTILITY button, then press the Arrow UP/Arrow DOWN button until you see the following display:

```
Lockout:
Off
```

Toggle the Encoder to select ON.

```
> Lockout     <
On
```

When set to ON, the User Lockout feature locks the current preset and prevents users from accessing the following buttons and their associated parameters:

PRESET
DELAY
LIMITER
UTILITY

Pressing any of these buttons or turning the Encoder wheel will yield the following message on the display:

```
> Lockout:    <
> No Access   <
```

To Disable User Lockout:

Press and hold the UTILITY button for five seconds until you see the following display:

```
> Lockout      <
> Unlocked     <
```

User Lockout is now disabled.

9.0 Router Functionality

The 2 x 4 router is updated every time a new preset is loaded to reflect the stored output configuration. The router can be manually edited by the user to accommodate additional output configurations.

To access the router, press the UTILITY button and use the Arrow UP/Arrow DOWN buttons to navigate through the Utility row. Use the Encoder wheel to change router values.The following routing parameters can be accessed under the UTILITY button:

Output Route 1:
Selects audio information from the CH1 or CH2 inputs and sends it to Output CH1.

Default Value: The routing information is stored with each preset.

Output Route 2:
Selects the type of audio information from the CH1 or CH2 inputs and sends it to Output CH2.

Default Value: The routing information is stored with each preset.

Output Route 3:
Selects the type of audio information from the CH1 or CH2 inputs and sends it to Output CH3.

Default Value: The routing information is stored with each preset.

Output Route 4:
Selects the type of audio information from the CH1 or CH2 inputs and sends it to Output CH4.

Default Value: The routing information is stored with each Preset.

Each of the four outputs can be set to the following:

1. **CH1/High Frequency:** Selects the high frequency information from the CH1 input.
2. **CH2/High Frequency:** Selects the high frequency information from the CH2 input.
3. **Mono/High Frequency:** Selects the high frequency information from the CH1 and CH2 inputs and sums them to Mono.
4. **CH1/Low Frequency:** Selects the low-frequency information from the CH1 input.
5. **CH2/Low Frequency:** Selects the low-frequency information from the CH2 Input.
6. **Mono/Low Frequency:** Selects the low-frequency information from the CH1 and CH2 inputs and sums them to Mono.
7. **CH1/Full Range:** Selects the low-frequency information from the CH1 input.
8. **CH2/Full Range:** Selects the low-frequency information from the CH2 input.
9. **Mono/Full Range:** Selects information from the CH1 and CH2 inputs and sums them to Mono.
10. **CH1/No EQ:** Bypasses active Bose® EQ for CH1 Program Material.
11. **CH2/No EQ:** Bypasses Active Bose EQ for CH2 Program Material.
12. **Mono/No EQ:** Sums the CH1 and CH2 to Mono and bypasses the Bose EQ presets.

Router Functionality

PANARAY® System Digital Controller Output Configuration Table

Router Matrix

Output Route 1	Output Route 2	Output Route 3	Output Route 4
CH1/High	CH1/High	CH1/High	CH1/High
CH2/High	CH2/High	CH2/High	CH2/High
Mono/High	Mono/High	Mono/High	Mono/High
CH1/Low	CH1/Low	CH1/Low	CH1/Low
CH2/Low	CH2/Low	CH2/Low	CH2/Low
Mono/Low	Mono/Low	Mono/Low	Mono/Low
CH1/Full Range	CH1/Full Range	CH1/Full Range	CH1/Full Range
CH2/Full Range	CH2/Full Range	CH2/Full Range	CH2/Full Range
Mono/Full Range	Mono/Full Range	Mono/Full Range	Mono/Full Range
CH1/No EQ	CH1/No EQ	CH1/No EQ	CH1/No EQ
CH2/No EQ	CH2/No EQ	CH2/No EQ	CH2/No EQ
Mono/No EQ	Mono/No EQ	Mono/No EQ	Mono/No EQ

10.0 Optimizing Signal-to-Noise

Optimizing Signal-to-Noise using the PANARAY® System Digital Controller with a Third-Party Amplifier

In order to achieve the best signal-to-noise performance with the PANARAY® system digital controller, you must optimize the input sensitivity and output range to match the equipment you are interfacing with.

1. Before making any connections between the PANARAY system digital controller and an amplifier, make sure all of your electronics are set to Off, and your amplifier level controls set to Minimum.

2. Next, connect an audio source to the PANARAY system digital controller inputs CH1 and CH2. If you have a mono source, plug your connection into CH1 and set the input configuration (located under the utility button) to CH1 (mono).

3. Connect the PANARAY system digital controller outputs to your amplifier inputs.

4. Turn on your audio source. Then turn on your PANARAY system digital controller.

5. Select a Speaker EQ on the Panaray controller by pressing the PRESET button and use the Encoder wheel to select the desired speaker EQ.

6. Set the PANARAY system digital controller's input sensitivity. In general you should set this parameter to match the output range of the equipment you connect to the controller. A digital product has the best signal-to-noise performance when operating at 0dBFS (full scale), so you should run the PANARAY system digital controller as close to 0dBFS as possible.

For example: If you have a mixer with +4dBU outputs, the PANARAY controller input sensitivity should be set to +8dBu. This will allow you to reach 0dBFS while at the same time providing some additional headroom to account for peaks in program material. Set the parameter based on the amount of headroom needed.

To access the PANARAY system digital controller input sensitivity, press the UTILITY button. Then press the Arrow DOWN button until you see the following display:

```
Input Sense
20dBu
```

Adjust this parameter with the Encoder wheel to match your audio source's maximum output.

```
Input Sense
8dBu
```

Next, send a signal from your audio source to the PANARAY system digital controller and view the front panel input meters to ensure your signal is at approximately 0dbFS (full scale). Adjust the input sensitivity parameter accordingly.

Note: The input signal may occasionally light the red Overload LED. This is normal.

7. After you have properly calibrated your PANARAY controller's input sensitivity, you must adjust your controller's output range to match your amplifier's input sensitivity. Since this will vary with each manufacturer, please consult your amplifier user guide.

10.0 Optimizing Signal-to-Noise

Use the table below to match the PANARAY® system digital controller's Output Range to your amplifier.

Output Range Parameter	**Amplifier Input Sensitivity**
2dBu	Use for anything less than 0.775V
8dBu	0.775V to 0.975V
14dBu	0.975V to 3.9V
20dBu	3.9V to 7.75V
Note: *If there are no settings that exactly match your amplifier's input sensitivity, pick the highest setting that is nearest your amplifier input sensitivity and use the controller's OUTPUT LEVEL control to scale back the overall output.*	

Set the PANARAY system digital controller output range. First, press the UTILITY button. Then press the Arrow DOWN button until you see the following display:

```
Output Range
20dB (Pro)
```

Adjust this parameter with the Encoder wheel to match your amplifier's input sensitivity.

```
Output Range
14dB (Consumer)
```

This setting will provide the optimum input level for your amplifier.

8. Select a speaker EQ preset on the PANARAY system digital controller by pressing the PRESET button and using the Encoder wheel to select the desired speaker EQ preset.

9. Turn your amplifier On, and turn up each level control until you reach the desired SPL.

These settings will automatically be retained.

11.0 Custom Preset Mode

The PANARAY® controller's Custom Preset feature allows users to edit and store the delay, limiting and routing settings. Up to 20 custom presets can be stored using this feature.

After making the desired changes to a preset, follow the steps below to store a new preset.

1. Press and hold the DELAY and LIMITER buttons at the same time (for approximately 5 seconds) until you see the following display:

   ```
   Store: Select no
   1>EMPTY PRESET
   ```

   Use the Encoder wheel to select the location where the preset will be stored (up to 20 locations can be used).

2. Once you have selected your location, press the Arrow DOWN button to name your preset. You will see the following display:

   ```
   Store: Edit name
   1 402I
   ```

   Use the Arrow UP/Arrow DOWN buttons to align the cursor under the desired letter or number you want to change, then move the Encoder wheel through the list of characters. Press the Arrow UP button to move to the next character, and press the Arrow DOWN button to move back. To CANCEL the store process, press the PRESET button.

3. Once you have named your preset, press and holdD the DELAY and LIMITER buttons until you see the following display:

   ```
   > Preset     <
   > Stored     <
   ```

After you store a program, you will be immediately transferred to the custom preset mode. The stored preset will be displayed. Note that the bank designation will be based on the preset that you edited.

For example: If the preset came from the Mono bank, it will be designated as Mono Custom. If a preset originates from the Stereo bank then it will be designated as Stereo Custom. And if a preset originates from the Bass Array bank, it will be designated as Bass Array Custom.

```
Mono Custom
402I - TEST
```

In this mode, the user only has access to the custom presets. Select presets in the same manner as in the standard operating mode. The Encoder wheel selects presets and the Arrow UP/Arrow DOWN buttons select the desired bank.

To Turn Off Custom Presets:
To turn off custom presets and return to standard operation press the UTILITY button and use the Arrow UP/Arrow DOWN button to find the Custom Presets parameter and set it to Off. You will immediately be transferred to standard operating mode.

To Activate Custom Presets:
To return to custom preset mode, go back to the UTILITY row and turn the Custom Presets parameter to On. You will immediately be transferred to custom preset mode.To create additional custom presets, repeat the procedure above.
Note: You cannot store a Dual Mono preset, but can create one by storing two custom mono presets, which you have access to in custom preset mode. Follow standard operation instructions for loading Dual Mono presets.

12.0 Bass Array Presets

The following presets are available in the PANARAY® system digital controller for Bass Array Applications:

MB4 2xBS280Hz – The MB4 2x Broadside Bass Array preset provides 360 degree horizontal coverage with a 10 to 15dB suppression in energy going up and down from 80 – 200Hz. The crossover is set at 280Hz and is recommended for use with the LT Series.

MB4 2xBS180Hz – The MB4 2x Broadside Bass Array preset provides 360 degree horizontal coverage with a 10 to 15dB suppression in energy going up and down from 80 – 200Hz. The crossover is set at 180Hz and is recommended for use with MA12, MB4 and 02 Series speakers.

MB4 2xEF280Hz – The MB4 2x Endfire Bass Array preset provides 180 degree conical dispersion @ 160Hz. The crossover is set at 280Hz and is recommended for use with MA12, MB4 and LT Series speakers.

MB4 2xEF180Hz – The MB4 2x Endfire Bass Array preset provides 180 degree conical dispersion @ 160Hz. The crossover is set at 180Hz and is recommended for use with MA12, MB4 and 02 Series speakers.

MB4 4xEF280Hz – The MB4 4x Endfire Bass Array preset provides 120 degree conical dispersion @ 160Hz. The crossover is set at 280Hz and is recommended for use with MA12, MB4 and LT Series speakers.

MB4 4xEF180Hz – The MB4 4x Endfire Bass Array preset provides 120 degree conical dispersion @ 160Hz. The crossover is set at 180Hz and is recommended for use with MA12, MB4 and 02 Series speakers.

Loading and Setting Up Bass Arrays Using Presets

1. Press the PRESET button on the front panel of the PANARAY system digital controller.

2. Press the Arrow DOWN button until you see the following display:

```
Bass Array
MB4 2xBS280Hz
```

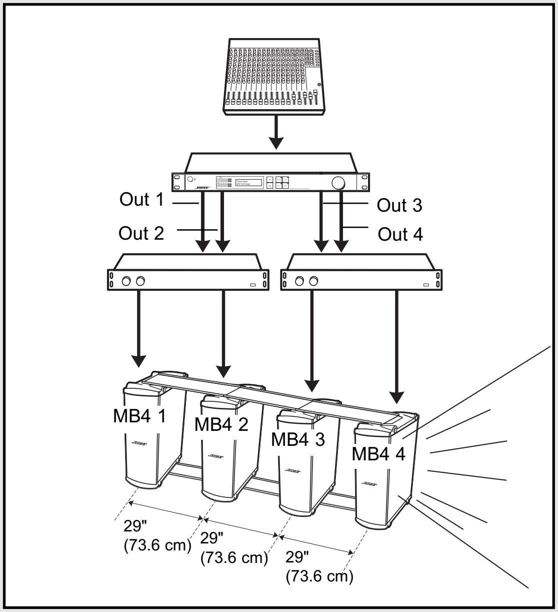
3. Turn the Encoder wheel to the right to select the desired bass array configuration.

```
Bass Array
MB4 4xEF280Hz
```

4. Once you have selected your bass array preset, the program is automatically loaded and ready to go.

12.1 Bass Array Setup and Connections

MB4 4x Endfire Bass Array



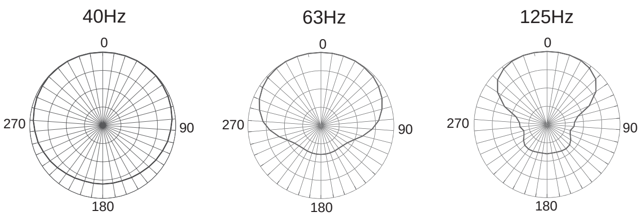
Bose® Components:
1 PANARAY® system digital controller
2 Bose 1800-VI power amplifiers
2 4x endfire bass array brackets (top and bottom)
4 MB4 modular bass loudspeakers

Spacing:
In order to effectively create the MB4 4x endfire bass array, each unit must be spaced 29" (73.6 cm) on center from each other. This is shown in the illustration above. IF THE UNITS ARE NOT SPACED PROPERLY THE BASS ARRAY WILL NOT WORK.

Note: All EQ and delay settings are already set in the PANARAY system digital controller. In order for the bass array to work, you must connect the MB4 units to the appropriate outputs on the controller and apply the proper spacing. The output assignments are listed to the right.

MB4 4x endfire bass array (280Hz and 180Hz) PANARAY® system digital controller output	
CH1	MB4 (1)
CH2	MB4 (2)
CH3	MB4 (3)
CH4	MB4 (4)

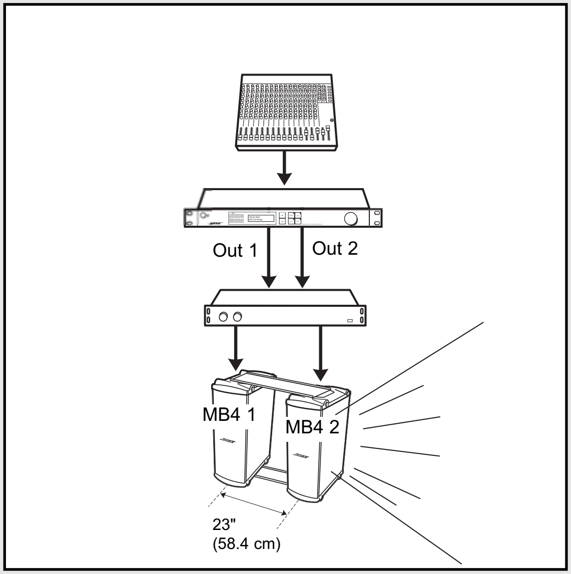
MB4 4x Endfire Horizontal Octave-Band Polar Data



This array fires from the end of the fourth MB4 speaker to provide the dispersion shown above.

12.1 Bass Array Setup and Connections

MB4 2x Endfire Bass Array



Bose® Components:

1 PANARAY® system digital controller
1 Bose 1800-VI power amplifier
2 2x endfire bass array brackets (top and bottom)
2 MB4 modular bass loudspeakers

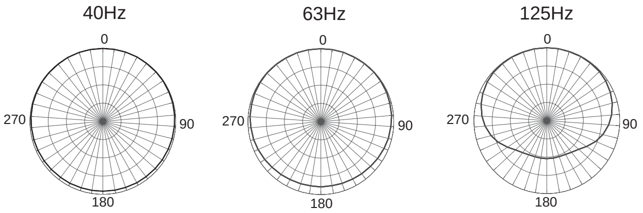
Spacing:

In order to effectively create the MB4 2x endfire bass array, each unit must be spaced 23" (58.4 cm) on center from each other). This is shown in the illustration above. IF THE UNITS ARE NOT SPACED PROPERLY THE BASS ARRAY WILL NOT WORK.

Note: All EQ and delay settings are already set in the PANARAY system digital controller. In order for the bass array to work, you must connect the MB4 units to the appropriate outputs on the controller and apply the proper spacing. The output assignments are listed to the right.

MB4 4x endfire bass array (280Hz and 180Hz) PANARAY system digital controller output	
CH1	MB4 (1)
CH2	MB4 (2)
CH3	MB4 (3)
CH4	MB4 (4)

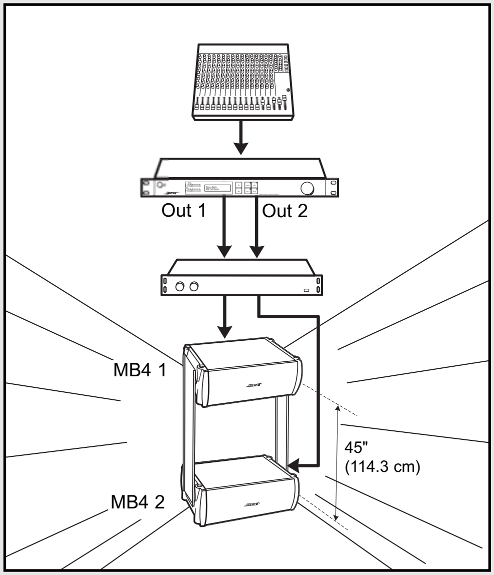
MB4 2x Endfire Horizontal Octave-Band Polar Data



This array fires from the end of the second MB4 speaker to provide the dispersion shown above.

12.1 Bass Array Setup and Connections

MB4 2x Broadside Bass Array



Bose® Components:

1 PANARAY® system digital controller
1 Bose 1800-VI power amplifier
2 2x broadside bass array brackets (top and bottom)
2 MB4 modular bass loudspeakers

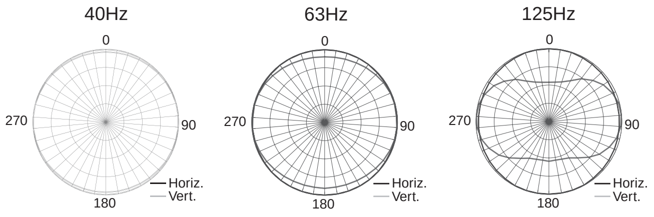
Spacing:

In order to effectively create the MB4 2x endfire bass array, each unit must be spaced 45" (114.3 cm) on center from each other. This is shown in the illustration above. IF THE UNITS ARE NOT SPACED PROPERLY THE BASS ARRAY WILL NOT WORK.

Note: All EQ and delay settings are already set in the PANARAY system digital controller. In order for the bass array to work, you must connect the MB4 units to the appropriate outputs on the controller and apply the proper spacing. The output assignments are listed to the right.

MB4 2x broadside bass array (280Hz and 180Hz) PANARAY® system digital controller output	
CH1	MB4 (1)
CH2	MB4 (2)
CH3	Not used
CH4	Not used

MB4 2x Broadside Horizontal Octave-Band Polar Data



This array fires from the sides of both MB4 speakers to provide the dispersion shown above.

13.0 Bose® Speaker EQ Preset List

The following is a list of 180 presets that are available in the PANARAY® system digital controller. The presets are separated into three separate banks: Mono, Stereo and Bass Array. Included in this list is the name of each preset as it appears on the PANARAY controller LCD as well as a description of the preset and the default settings for each of the four outputs.

LCD Display	Preset Description	Output Default Configuration			
Mono Presets	**Preset Description**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
Mono Preset Name	Preset Description	Preset Default	Preset Default	Preset Default	Preset Default
Stereo Preset	**Preset Description**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
Stereo Preset Name	Preset Description	Preset Default	Preset Default	Preset Default	Preset Default
Bass Array	**Preset Description**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
Bass Array Preset Name	Preset Description	Preset Default	Preset Default	Preset Default	Preset Default

13.1 Bose® Speaker EQ Preset List – Mono Bank

PANARAY 402 Series I loudspeaker Mono Presets	402 Series I loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
402I	402 Series I loudspeaker Full Range EQ	402I Mono Full Range	402I Mono Full Range	402I Mono Full Range	402I Mono Full Range
402I+502B	402 Series I loudspeaker bi-amped with 502B loudspeaker	402I Mono High	402I Mono High	502B Mono Low	502B Mono Low
402I+MB4	402 Series I loudspeaker bi-amped with MB4	402I Mono High	402I Mono High	MB4 Mono Low	MB4 Mono Low
402I+AWCS	402 Series I loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	402I Mono High	402I Mono High	AWCS Mono Low	AWCS Mono Low
402I HF	402 Series I loudspeaker bi-amped @ 160Hz	402I Mono High	402I Mono High	402I Mono High	402I Mono High

PANARAY 402 Series II loudspeaker Mono Presets	402 Series II loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
402II	402 Series II loudspeaker Full Range EQ	402II Mono Full Range	402II Mono Full Range	402II Mono Full Range	402II Mono Full Range
402II+502B	402 Series II loudspeaker bi-amped with 502B loudspeaker	402II Mono High	402II Mono High	502B Mono Low	502B Mono Low
402II+502BEX	402 Series II loudspeaker bi-amped with 502BEX loudspeaker	402II Mono High	402II Mono High	502BEX Mono Low	502BEX Mono Low
402II+MB4	402 Series II loudspeaker bi-amped with MB4	402II Mono High	402II Mono High	MB4 Mono Low	MB4 Mono Low
402II+AWCS	402 Series II loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS.	402II Mono High	402II Mono High	AWCS Mono Low	AWCS Mono Low
402II HF	402 Series II loudspeaker bi-amped @ 180Hz	402II Mono High	402II Mono High	402II Mono High	402II Mono High

13.1 Bose® Speaker EQ Preset List – Mono Bank

PANARAY® 502® A loudspeaker Mono Presets	502A loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
502A	502A loudspeaker Full Range EQ	502A Mono Full Range	502A Mono Full Range	502A Mono Full Range	502A Mono Full Range
502A+502B	502A loudspeaker bi-amped with 502B	502A Mono High	502A Mono High	502B Mono Low	502B Mono Low
502A+MB4	502A loudspeaker bi-amped with MB4	502A Mono High	502A Mono High	MB4 Mono Low	MB4 Mono Low
502A+AWCS	502A loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	502A Mono High	502A Mono High	AWCS Mono Low	AWCS Mono Low
502A HF	502A loudspeaker bi--amped @ 160Hz	502A Mono High	502A Mono High	502A Mono High	502A Mono High

PANARAY 802® Series II loudspeaker Mono Presets	802 Series II loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
802II	802 Series II loudspeaker Full Range EQ	802II Mono Full Range	802II Mono Full Range	802II Mono Full Range	802II Mono Full Range
802II+502B	802 Series II loudspeaker bi-amped with 502B loudspeaker	802II Mono High	802II Mono High	502B Mono Low	502B Mono Low
802II+MB4	802 Series II loudspeaker bi-amped with MB4	802II Mono High	802II Mono High	MB4 Mono Low	MB4 Mono Low
802II+AWCS	802 Series II loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802II Mono High	802II Mono High	AWCS Mono Low	AWCS Mono Low
802II HF	802 Series II loudspeaker bi-amped @ 112Hz	802II Mono High	802II Mono High	802II High	802II High
802IISTK	Stacked 802 Series II loudspeaker EQ	802II Full Range	802II Full Range	802II Full Range	802II Full Range
802IISTK+502B	Stacked 802 Series II loudspeakers bi-amped with 502B loudspeaker	802II Mono High	802II Mono High	502B Mono Low	502B Mono Low

Bose® Speaker EQ Preset List – Mono Bank

PANARAY® 802® Series II loudspeaker Mono Presets	**802 Series II loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
802IISTK+AWCS	Stacked 802 Series II loudspeakers biamped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802II Mono High	802II Mono High	AWCS Mono Low	AWCS Mono Low
802IISTK HF	Stacked 802 Series II loudspeakers biamped @ 112Hz	802II Mono High	802II Mono High	802II Mono High	802II Mono High

PANARAY 802 Series III loudspeaker Mono Presets	**802 Series III loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
802III	802 Series III loudspeaker Full Range EQ	802III Mono Full Range	802III Mono Full Range	802III Mono Full Range	802III Mono Full Range
802III+502B	802 Series III loudspeaker bi-amped with 502B loudspeaker	802III Mono High	802III Mono High	502B Mono Low	502B Mono Low
802III+502BEX	802 Series III loudspeaker bi-amped with 502BEX loudspeaker	802III Mono High	802III Mono High	502BEX Mono Low	502BEX Mono Low
802III+MB4	802 Series III loudspeaker bi-amped with MB4	802III Mono High	802III Mono High	MB4 Mono Low	MB4 Mono Low
802III+AWCS	802 Series III loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802III Mono High	802III Mono High	AWCS Mono Low	AWCS Mono Low
802III HF	802 Series III loudspeaker bi-amped @ 125Hz	802III Mono High	802III Mono High	802III Mono High	802III Mono High
802IIIST	Stacked 802 Series III loudspeaker EQ	802III Mono Full Range	802III Mono Full Range	802III Mono Full Range	802III Mono Full Range
802IIIST+502B	Stacked 802 Series III loudspeakers bi-amped with 502B loudspeaker	802III Mono High	802III Mono High	502B Mono Low	502B Mono Low
802IIIST+502X	Stacked 802 Series III loudspeakers bi-amped with 502BEX loudspeaker	802III Mono High	802III Mono High	502BEX Mono Low	502BEX Mono Low
802IIISTK+MB4	Stacked 802 Series III loudspeakers bi-amped with MB4	802III Mono High	802III Mono High	MB4 Mono Low	MB4 Mono Low

Bose® Speaker EQ Preset List – Mono Bank

PANARAY® 802® Series III loudspeaker Mono Presets	**802 Series III loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
802IIISTK+AWCS	Stacked 802 Series III loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802III Mono/High	802III Mono/High	802III Mono/Low	802III Mono/Low
802IIISTK HP	Stacked 802 Series III loudspeaker bi-amped @ 125 Hz	802III Mono/High	802III Mono/High	802III Mono/High	802III Mono/High

PANARAY MA12 loudspeaker Mono Presets	**MA12 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
MA12 WALL	MA12 loudspeaker Full Range, Wall Mount, EQ Curve	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range
MA12 FREE	MA12 loudspeaker Full Range, Free Field, EQ Curve	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range
MA12+502B	MA12 loudspeaker bi-amped with 502®B loudspeaker	MA12 Mono High	MA12 Mono High	502B Mono Low	502B Mono Low
MA12+MB4	MA12 loudspeaker bi-amped with MB4	MA12 Mono High	MA12 Mono High	MB4 Mono Low	MB4 Mono Low
MA12 HF	MA12 loudspeaker bi-amped @ 160 Hz	MA12 Mono High	MA12 Mono High	MA12 Mono High	MA12 Mono High
MA12STK WALL	Stacked MA12 loudspeaker Full Range, Wall Mount, EQ Curve	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range
MA12STK FREE	Stacked MA12 loudspeaker Full Range, Free Field, EQ Curve	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range	MA12 Mono Full Range
MA12STK+502B	Stacked MA12 loudspeaker bi-amped with 502B loudspeaker	MA12 Mono High	MA12 Mono High	502B Mono Low	502B Mono Low
MA12STK+MB4	Stacked MA12 loudspeaker bi-amped with MB4	MA12 Mono High	MA12 Mono High	MB4 Mono Low	MB4 Mono Low
MA12STK HF	Stacked MA12 loudspeaker bi-amped @ 160Hz	MA12 Mono High	MA12 Mono High	MA12 Mono High	MA12 Mono High

Bose® Speaker EQ Preset List – Mono Bank

PANARAY® MB4 loudspeaker Mono Presets	MB4 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
MB4 100Hz LP	MB4 loudspeaker 100Hz Low Pass	MB4	MB4	MB4	MB4
MB4 160Hz LP	MB4 loudspeaker 160Hz Low Pass	MB4	MB4	MB4	MB4
MB4 200Hz LP	MB4 loudspeaker 200Hz Low Pass	MB4	MB4	MB4	MB4
MB4 280Hz LP	MB4 loudspeaker 280Hz Low Pass	MB4	MB4	MB4	MB4

PANARAY 502® B loudspeaker Mono Presets	502B loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
502B	502B loudspeaker Mono Low Pass @ 160 Hz	502B	502B	502B	502B

PANARAY 502® BEX loudspeaker Mono Presets	502BEX loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
502BEX	502BEX loudspeaker Mono Low Pass @ 200 Hz	502BEX	502BEX	502BEX	502BEX

AWCS Mono Presets	AWCS Preset Descriptions	Output 1	Output 2	Output 3	Output 4
AWCS	AWCS Mono Low Pass with limiting assigned to all four outputs	AWCS	AWCS	AWCS	AWCS

PANARAY LT3202® Mono Presets	LT3202 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
LT3202	LT3202 loudspeaker Full Range EQ	LT3202 Mono Full Range	LT3202 Mono Full Range	LT3202 Mono Full Range	LT3202 Mono Full Range
LT3202CLUSTER	LT3202 loudspeaker Cluster EQ	LT3202 Mono Full Range	LT3202 Mono Full Range	LT3202 Mono Full Range	LT3202 Mono Full Range
LT3202+MB4	LT3202 loudspeaker bi-amped with MB4	LT3202 Mono High	LT3202 Mono High	MB4 Mono Low	MB4 Mono Low
LT3202+502BEX	LT3202 loudspeaker bi-amped with 502BEX loudspeaker	LT3202 Mono High	LT3202 Mono High	502BEX Mono Low	502BEX Mono Low

Bose® Speaker EQ Preset List – Mono Bank

PANARAY LT4402 loudspeaker Mono Presets	**LT4402 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
LT4402	LT4402 loudspeaker Full Range EQ	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range
LT4402CLUSTER	LT4402 Cluster EQ	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range
LT4402+MB4	LT4402 loudspeaker bi-amped with MB4	LT4402 Mono High	LT4402 Mono High	MB4 Mono Low	MB4 Mono Low
LT4402+502BEX	LT4402 loudspeaker bi-amped with 502 BEX loudspeaker	LT4402 Mono High	LT4402 Mono High	502BEX Mono Low	502BEX Mono Low

PANARAY LT9702 loudspeaker Mono Presets	**LT9702 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
LT9702	LT9702 loudspeaker Full Range EQ	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range
LT9702CLUSTER	LT9702 Cluster EQ	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range
LT9702+MB4	LT9702 loudspeaker bi-amped with MB4	LT9702 Mono High	LT9702 Mono High	MB4 Mono Low	MB4 Mono Low
LT9702+502BEX	LT9702 loudspeaker bi-amped with 502BEX loudspeaker	LT9702 Mono High	LT9702 Mono High	502BEX Mono Low	502BEX Mono Low

FreeSpace Model 8 loudspeaker Mono Presets	**Model 8 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
M8	Model 8 loudspeaker Full Range EQ	M8 Mono Full Range	M8 Mono Full Range	M8 Mono Full Range	M8 Mono Full Range
M8+MB4	Model 8 loudspeaker bi-amped with MB4	M8 Mono High	M8 Mono High	MB4 Mono Low	MB4 Mono Low
M8+FS3	Model 8 loudspeaker bi-amped with FreeSpace 3 loudspeaker	M8 Mono High	M8 Mono High	FS3 Mono Low	FS3 Mono Low
M8+FS1B	Model 8 loudspeaker bi-amped with FreeSpace 1B loudspeaker	M8 Mono High	M8 Mono High	FS1B Mono Low	FS1B Mono Low

Bose® Speaker EQ Preset List – Mono Bank

FreeSpace Model 32 loudspeaker Mono Presets	Model 32 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
M32	Model 32 loudspeaker Full Range EQ	M32 Mono Full Range	M32 Mono Full Range	M32 Mono Full Range	M32 Mono Full Range
M32+MB4	Model 32 loudspeaker bi-amped with MB4	M32 Mono High	M32 Mono High	MB4 Mono Low	MB4 Mono Low
M32+FS3	Model 32 loudspeaker bi-amped with FS3	M32 Mono High	M32 Mono High	FS3 Mono Low	FS3 Mono Low
M32+FS1B	Model 32 loudspeaker bi-amped with FS1B	M32 Mono High	M32 Mono High	FS1B Low	FS1B Low
M32+502BEX	Model 32 loudspeaker bi-amped with 502 BEX loudspeaker	M32 Mono High	M32 Mono High	502BEX Mono Low	502BEX Mono Low

FreeSpace Model 1B loudspeaker Mono Presets	FreeSpace Model 1B loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
FS1B 100Hz LP	FreeSpace Model 1B loudspeaker 100Hz Low Pass	FS1B	FS1B	FS1B	FS1B
FS1B SURFACE	FreeSpace Model 1B loudspeaker Surface Mount EQ	FS1B	FS1B	FS1B	FS1B
FS1B FLUSH	FreeSpace Model 1B loudspeaker Flush Mount EQ	FS1B	FS1B	FS1B	FS1B

FreeSpace 3 Bass Mono Presets	FreeSpace 3 Bass Preset Descriptions	Output 1	Output 2	Output 3	Output 4
FS3 100Hz LP	FreeSpace 3 Bass 100Hz Low Pass	FS3	FS3	FS3	FS3
FS3 150Hz LP	FreeSpace 3 Bass 150Hz Low Pass	FS3	FS3	FS3	FS3

FreeSpace 360 Mono Presets	FreeSpace 360 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
FS360 HARD	FS360 loudspeaker inground hard EQ	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range
FS360 SOFT	FS360 loudspeaker inground soft EQ	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range
FS360 DECK	FS360 loudspeaker deck EQ	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range

13.2 Bose® Speaker EQ Preset List – Stereo Bank

PANARAY 402 Series I loudspeaker Stereo Presets	**402 Series I loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
402I	402 Series I loudpeaker Full Range EQ	402I CH1/Full Range	402I CH2/Full Range	402I CH1/Full Range	402I CH2/Full Range
402I+502B	402 Series I loudspeaker bi-amped with 502 B loudspeaker	402I CH1/High	402I CH2/High	502B CH1/Low	502B CH2/Low
402I+MB4	402 Series I loudspeaker bi-amped with MB4	402I CH1/High	402I CH2/High	MB4 CH1/Low	MB4 CH2/Low
402I+AWCS	402 Series I loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	402I CH1/High	402I CH2/High	AWCS CH1/Low	AWCS CH2/Low
402I HF	402 Series I loudspeaker bi-amped @ 160Hz	402I CH1/High	402I CH2/High	402I CH1/High	402i CH2/High

Panaray 402 Series II loudspeaker Stereo Presets	**402 Series II loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
402II	402 Series II loudpeaker Full Range EQ	402II CH1/Full Range	402II CH2/Full Range	402II CH1/Full Range	402II CH2/Full Range
402II+502B	402 Series II loudspeaker bi-amped with 502B loudspeaker	402II CH1/High	402II CH2/High	502B CH1/Low	502B CH2/Low
402II+502BEX	402 Series II loudspeaker bi-amped with 502 BEX loudspeaker	402II CH1/High	402II CH2/High	502BEX CH1/Low	502BEX CH2/Low
402II+MB4	402 Series II loudspeaker bi-amped with MB4	402II CH1/High	402II CH2/High	MB4 CH1/Low	MB4 CH2/Low
402II+AWCS	402 Series II loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	402II CH1/High	402II CH2/High	AWCS CH1/Low	AWCS CH2/Low
402II HF	402 Series II loudspeaker bi-amped @ 180Hz	402II CH1/High	402II CH2/High	402II CH1/High	402II CH2/High

Bose® Speaker EQ Preset List – Stereo Bank

PANARAY□ 502□ A loudspeaker Stereo Presets	**502A loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
502A	502A loudspeaker Full Range EQ	502A CH1/Full Range	502A CH2/Full Range	502A CH1/Full Range	502A CH2/Full Range
502A+502B	502A loudspeaker bi-amped with 502B	502A CH1/High	502A CH2/High	502B CH1/Low	502B CH2/Low
502A+MB4	502A loudspeaker bi-amped with MB4	502A CH1/High	502A CH2/High	MB4 CH1/Low	MB4 CH2/Low
502A+AWCS	502A loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	502A CH1/High	502A CH2/High	AWCS CH1/Low	AWCS CH2/Low
502A HF	502A loudspeaker bi-amped @ 160Hz	502A CH1/High	502A CH2/High	502A CH1/High	502A CH2/High

PANARAY 802□ Series II loudspeaker Stereo Presets	**802 Series II loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
802II	802 Series II loudspeaker Full Range EQ	802II CH1/Full Range	802II CH2/Full Range	802II CH1/Full Range	802II CH2/Full Range
802II+502B	802 Series II loudspeaker bi-amped with 502B loudspeaker	802II CH1/High	802II CH2/High	502B CH1/Low	502B CH2/Low
802II+MB4	802 Series II loudspeaker bi-amped with MB4	802II CH1/High	802II CH2/High	MB4 CH1/Low	MB4 CH2/Low
802II+AWCS	802 Series II loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802II CH1/High	802II CH2/High	AWCS CH1/Low	AWCS CH2/Low
802II HF	802 Series II loudspeaker bi-amped @ 112Hz	802II CH1/High	802II CH2/High	802II CH1/High	802II CH2/High
802IISTK	Stacked 802 Series II loudspeaker EQ	802II CH1/Full Range	802II CH2/Full Range	802II CH1/Full Range	802II CH2/Full Range
802IISTK+502B	Stacked 802 Series II loudspeakers bi-amped with 502B loudpeaker	802II CH1/High	802II CH2/High	502B CH1/Low	502B CH2/Low
802IISTK+MB4	Stacked 802 Series II loudspeakers bi-amped with MB4	802II CH1/High	802II CH2/High	MB4 CH1/Low	MB4 CH2/Low

13.2 Bose® Speaker EQ Preset List – Stereo Bank

PANARAY□ 802□ Series II loudspeaker Stereo Presets	802 Series II loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
802IISTK+AWCS	Stacked 802 Series II loudspeakers bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802II CH1/High	802II CH2/High	AWCS CH1/Low	AWCS CH2/Low
802IISTK HF	Stacked 802 Series II loudspeakers bi-amped @ 112Hz	802II CH1/High	802II CH2/High	802II CH1/High	802II CH2/High

PANARAY 802□ Series III loudspeaker Stereo Presets	802 Series III loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
802III	802 Series III loudspeaker Full Range EQ	802III CH1/Full Range	802III CH2/Full Range	802III CH1/Full Range	802III CH2/Full Range
802III+502B	802 Series III loudspeaker bi-amped with 502□ B loudspeaker	802III CH1/High	802III CH2/High	502B CH1/Low	502B CH2/Low
802III+502BEX	802 Series III loudspeaker bi-amped with 502BEX loudspeaker	802III CH1/High	802III CH2/High	502BEX CH1/Low	502BEX Ch2/Low
802III+MB4	802 Series III loudspeaker bi-amped with MB4	802III CH1/High	802III CH2/High	MB4 CH1/Low	MB4 CH2/Low
802III+AWCS	802 Series III loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802III CH1/High	802III CH2/High	AWCS CH1/Low	AWCS CH2/Low
802III HF	802 Series III loudspeaker bi-amped @ 125Hz	802III CH1/High	802III CH2/High	802III CH1/High	802III CH2/High
802IIIST	Stacked 802 Series III loudspeaker EQ	802III CH1/Full Range	802III CH2/Full Range	802III CH1/Full Range	802III CH2/Full Range
802IIIST+502B	Stacked 802 Series III loudspeakers bi-amped with 502B	802III CH1/High	802III CH2/High	502B CH1/Low	502B CH2/Low
802IIIST+502X	Stacked 802 Series III loudspeakers bi-amped with 502BEX	802III CH1/High	802III CH2/High	502BEX CH1/Low	502BEX CH2/Low
802IIISTK+MB4	Stacked 802 Series III loudspeakers bi-amped with MB4	802III CH1/High	802III CH2/High	MB4 CH1/Low	MB4 CH2/Low

Bose® Speaker EQ Preset List – Stereo Bank

PANARAY® 802® Series III loudspeaker Stereo Presets	**802 Series III loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
802IIISTK+AWCS	Stacked 802 Series III loudspeaker bi-amped with AWCS Note: Limiting is assigned to outputs 3 and 4 for protection of the AWCS	802III CH1/High	802III CH2/High	AWCS CH1/Low	AWCS CH2/Low
802IIISTK HP	Stacked 802 Series III loudspeaker bi-amped @ 125 Hz	802III CH1/High	802III CH2/High	802III CH1/High	802III CH2/High

PANARAY MA12 loudspeaker Stereo Presets	**MA12 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
MA12 WALL	MA12 loudspeaker Full Range, Wall Mount, EQ Curve	MA12 CH1/Full Range	MA12 CH2/Full Range	MA12 CH1/Full Range	MA12 CH2/Full Range
MA12 FREE	MA12 loudspeaker Full Range, Free Field, EQ Curve	MA12 CH1/Full Range	MA12 CH2/Full Range	MA12 CH1/Full Range	MA12 CH2/Full Range
MA12+502B	MA12 loudspeaker bi-amped with 502®B loudspeaker	MA12 CH1/High	MA12 CH2/High	502B CH1/Low	502B CH2/Low
MA12+MB4	MA12 loudspeaker bi-amped with MB4	MA12 CH1/High	MA12 CH2/High	MB4 CH1/Low	MB4 CH2/ Low
MA12 HF	MA12 loudspeaker bi-amped @ 160 Hz	MA12 CH1/High	MA12 CH2/High	MA12 Mono High	MA12 Mono High
MA12STK WALL	Stacked MA12 loudspeaker Full Range, Wall Mount, EQ Curve	MA12 CH1/Full Range	MA12 CH2/Full Range	MA12 CH1/Full Range	MA12 CH2/Full Range
MA12STK FREE	Stacked MA12 loudspeaker Full Range, Free Field, EQ Curve	MA12 CH1/Full Range	MA12 CH2/Full Range	MA12 CH1/Full Range	MA12 CH2/Full Range
MA12STK+502B	Stacked loudspeaker bi-amped with 502B loudspeaker	MA12 CH1/High	MA12 CH2/High	502B CH1/Low	502B CH2/Low
MA12STK+MB4	Stacked MA12 loudspeaker bi-amped with MB4	MA12 CH1/High	MA12 CH2/High	MB4 CH1/Low	MB4 CH2/Low
MA12STK HF	Stacked MA12 loudspeaker bi-amped @ 160Hz	MA12 CH1/High	MA12 CH2/High	MA12 CH1/High	MA12 CH2/High

Bose® Speaker EQ Preset List – Stereo Bank

PANARAY® MB4 loudspeaker Stereo Presets	**MB4 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
MB4 100Hz LP	MB4 loudspeaker 100Hz Low Pass	MB4/CH1	MB4/CH2	MB4/CH1	MB4/CH2
MB4 160Hz LP	MB4 loudspeaker 160Hz Low Pass	MB4/CH1	MB4/CH2	MB4/CH1	MB4/CH2
MB4 200Hz LP	MB4 loudspeaker 200Hz Low Pass	MB4/CH1	MB4/CH2	MB4/CH1	MB4/CH2
MB4 280Hz LP	MB4 loudspeaker 280Hz Low Pass	MB4/CH1	MB4/CH2	MB4/CH1	MB4/CH2

PANARAY 502® B loudspeaker Mono Presets	**502B loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
502B	502B loudspeaker Mono Low Pass @ 160 Hz	502B/CH1	502B/CH2	502B/CH1	502B/CH2

PANARAY 502® BEX loudspeaker Stereo Presets	**502BEX loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
502BEX	502BEX loudspeaker Mono Low Pass @ 200 Hz	502BEX/CH1	502BEX/CH2	502BEX/CH1	502BEX/CH2

AWCS Stereo Presets	**AWCS Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
AWCS	AWCS Mono Low Pass with limiting assigned to all four outputs	AWCS/CH1	AWCS/CH2	AWCS/CH1	AWCS/CH2

PANARAY LT3202® Stereo Presets	**LT3202 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
LT3202	LT3202 loudspeaker Full Range EQ	LT3202 CH1 Full Range	LT3202 CH2 Full Range	LT3202 CH1 Full Range	LT3202 CH2 Full Range
LT3202CLUSTER	LT3202 loudspeaker Cluster EQ	LT3202 CH1 Full Range	LT3202 CH2 Full Range	LT3202 CH1 Full Range	LT3202 CH2 Full Range
LT3202+MB4	LT3202 loudspeaker bi-amped with MB4	LT3202 CH1/High	LT3202 CH2/High	MB4 CH1/Low	MB4 CH2/Low
LT3202+502BEX	LT3202 loudspeaker bi-amped with 502BEX loudspeaker	LT3202 CH1/High	LT3202 CH2/High	502BEX CH1/Low	502BEX CH2/Low

Bose® Speaker EQ Preset List – Stereo Bank

PANARAY□ LT4402□ loudspeaker Mono Presets	LT4402 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
LT4402	LT4402 loudspeaker Full Range EQ	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range
LT4402CLUSTER	LT4402 Cluster EQ	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range	LT4402 Mono Full Range
LT4402+MB4	LT4402 loudspeaker bi-amped with MB4	LT4402 Mono High	LT4402 Mono High	MB4 Mono Low	MB4 Mono Low
LT4402+502BEX	LT4402 loudspeaker bi-amped with 502□BEX loudspeaker	LT4402 Mono High	LT4402 Mono High	502BEX Mono Low	502BEX Mono Low

PANARAY LT9702□ loudspeaker Mono Presets	LT9702 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
LT9702	LT9702 loudspeaker Full Range EQ	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range
LT9702CLUSTER	LT9702 Cluster EQ	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range	LT9702 Mono Full Range
LT9702+MB4	LT9702 loudspeaker bi-amped with MB4	LT9702 Mono High	LT9702 Mono High	MB4 Mono Low	MB4 Mono Low
LT9702+502BEX	LT9702 loudspeaker bi-amped with 502BEX loudspeaker	LT9702 Mono High	LT9702 Mono High	502BEX Mono Low	502BEX Mono Low

FreeSpace□ Model 8 loudspeaker Mono Presets	Model 8 loudspeaker Preset Descriptions	Output 1	Output 2	Output 3	Output 4
M8	Model 8 loudspeaker Full Range EQ	M8 Mono Full Range	M8 Mono Full Range	M8 Mono Full Range	M8 Mono Full Range
M8+MB4	Model 8 loudspeaker bi-amped with MB4	M8 Mono High	M8 Mono High	MB4 Mono Low	MB4 Mono Low
M8+FS3	Model 8 loudspeaker bi-amped with FreeSpace□ 3 loudspeaker	M8 Mono High	M8 Mono High	FS3 Mono Low	FS3 Mono Low
M8+FS1B	Model 8 loudspeaker bi-amped with FreeSpace 1B loudspeaker	M8 Mono High	M8 Mono High	FS1B Mono Low	FS1B Mono Low

Bose® Speaker EQ Preset List – Stereo Bank

FreeSpace® Model 32 loudspeaker Mono Presets	**Model 32 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
M32	Model 32 loudspeaker Full Range EQ	M32 Mono Full Range	M32 Mono Full Range	M32 Mono Full Range	M32 Mono Full Range
M32+MB4	Model 32 loudspeaker bi-amped with MB4	M32 Mono High	M32 Mono High	MB4 Mono Low	MB4 Mono Low
M32+FS3	Model 32 loudspeaker bi-amped with FS3	M32 Mono High	M32 Mono High	FS3 Mono Low	FS3 Mono Low
M32+FS1B	Model 32 loudspeaker bi-amped with FS1B	M32 Mono High	M32 Mono High	FS1B Low	FS1B Low
M32+502BEX	Model 32 loudspeaker bi-amped with 502® BEX loudspeaker	M32 Mono High	M32 Mono High	502BEX Mono Low	502BEX Mono Low

FreeSpace Model 1B loudspeaker Mono Presets	**FreeSpace Model 1B loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
FS1B 100Hz LP	FreeSpace Model 1B loudspeaker 100Hz Low Pass	FS1B	FS1B	FS1B	FS1B
FS1B SURFACE	FreeSpace Model 1B loudspeaker Surface Mount EQ	FS1B	FS1B	FS1B	FS1B
FS1B FLUSH	FreeSpace Model 1B loudspeaker Flush Mount EQ	FS1B	FS1B	FS1B	FS1B

FreeSpace 3 Bass Mono Presets	**FreeSpace 3 Bass Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
FS3 100Hz LP	FreeSpace 3 Bass 100Hz Low Pass	FS3	FS3	FS3	FS3
FS3 150Hz LP	FreeSpace 3 Bass 150Hz Low Pass	FS3	FS3	FS3	FS3

FreeSpace 360 Mono Presets	**FreeSpace 360 loudspeaker Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
FS360 HARD	FS360 loudspeaker inground hard EQ	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range
FS360 SOFT	FS360 loudspeaker inground soft EQ	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range
FS360 DECK	FS360 loudspeaker deck EQ	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range	FS360 Mono Full Range

Bose® Speaker EQ Preset List – Bass Array Bank

PANARAY MB4 Bass Array Presets	**MB4 Bass Array Preset Descriptions**	**Output 1**	**Output 2**	**Output 3**	**Output 4**
MB4 2xBS180Hz	The MB4 2x broadside bass array preset provides 360 degree horizontal coverage with a 10 to 15 dB suppression in energy going up and down from 80 - 200Hz. The crossover is set at 180Hz and is recommended for use with MA12, MB4 and 02 Series speakers.	MB4/Mono	MB4/Mono	N/A	N/A
MB4 2xBS280Hz	The MB4 2x broadside bass array preset provides 360 degree horizontal coverage with a 10 to 15 dB suppression in energy going up and down from 80 - 200Hz. The crossover is set at 280Hz and is recommended for use with the LT Series.	MB4/Mono	MB4/Mono	N/A	N/A
MB4 2xEF180Hz	The MB4 2x endfire bass array preset provides 180 degree conical dispersion @ 160Hz. The crossover is set at 180Hz and is recommended for use with MA12, MB4 and 02 Series speakers.	MB4/Mono	MB4/Mono	N/A	N/A
MB4 2xEF280Hz	The MB4 2x endfire bass array preset provides 180 degree conical dispersion @ 160Hz. The crossover is set at 280Hz and is recommended for use with MA12, MB4 and LT Series speakers.	MB4/Mono	MB4/Mono	N/A	N/A
MB4 4xEF280Hz	The MB4 4x endfire bass array preset provides 120 degree conical dispersion @ 160Hz. The crossover is set at 280Hz and is recommended for use with MA12, MB4 and LT Series speakers.	MB4/Mono	MB4/Mono	MB4/Mono	MB4/Mono
MB4 4xEF180Hz	The MB4 4x endfire bass array preset provides 120 degree conical dispersion @ 160Hz. The crossover is set at 180Hz and is recommended for use with MA12, MB4 and 02 Series speakers.	MB4/Mono	MB4/Mono	MB4/Mono	MB4/Mono

14.0 Technical Information

Sample Rate
44.1kHz

Analog Inputs:

Connectors
(2) XLR balanced

Impedance balanced
21kΩ

Input level
Maximum: +18dBu
Minimum: 0dBu

Sensitivity
@9dB headroom: -12dBu to +9dBu

A to D conversion
24 bit, 128x oversampling bitstream

Dynamic range
95dB typ, 20Hz - 20kHz

THD
Typ < 92dB (0.0025%) @ 1kHz

Frequency response
20Hz to 20kHz, +0/-0.1dB

Crosstalk
< -100dB, 20Hz to 20kHz

Analog Outputs:

Connectors
(4) XLR balanced

Impedance balanced
40Ω

Maximum output level
+18dBu (balanced)

Output ranges
Balanced 20/14/8/2dBu

D to A conversion
24 bit, 128x oversampling bitstream

D to A Delay
0.63ms @ 44.1kHz

Dynamic range
100dB typ 20Hz - 20kHz

THD
Typ < 94dB (0.002%) @ 1kHz +20dBu output

Frequency response
20Hz to 20kHz: +0/-0.5dB

Crosstalk
< -95dB, 20Hz to 20kHz

Environment:

Operating temperature
32˚F to 122˚F (0˚C to 50˚C)

Storage temperature
-22˚F to 167˚F (-30˚C to 70˚C)

Humidity
Maximum 90% non-condensing

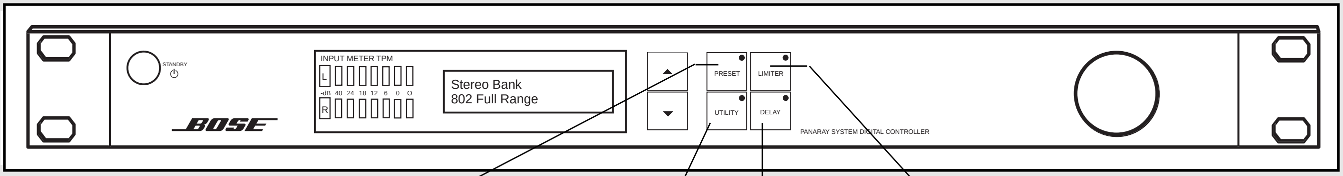
Control interface:

Finish
Black anodized aluminum front-plated and painted steel chassis

Mains voltage
100 to 240VAC, 50Hz to 60Hz (auto switching)

Power consumption
< 15W

Appendix: PANARAY® System Digital Controller Quick-Reference Guide


STANDBY
BOSE
INPUT METER TPM
L
-dB 40 24 18 12 6 0 O
R
Stereo Bank
802 Full Range
PRESET
LIMITER
UTILITY
DELAY
PANARAY SYSTEM DIGITAL CONTROLLER

PRESET Button

180 Speaker EQ presets sorted into Mono, Stereo, and Bass Array Banks

UTILITY Button

Output level
Output range
Input sensitivity
Input trim CH1
Input trim CH2
Output route 1
Output route 2
Output route 3
Output route 4
Delay units
Delay master
Limiter master
Input configuration
Custom presets
Input configuration
Custom presets
View angle

DELAY Button

Delay 1 On/Off
Delay 2 On/Off
Delay 3 On/Off
Delay 4 On/Off
Delay Time 1
Delay Time 2
Delay Time 3

LIMITER Button

Lim1 On/Off
Lim2 On/Off
Lim3 On/Off
Lim4 On/Off
Lim1 Threshold (L1)
Lim1 Ratio (L1)
Lim1 Attack (L1)
Lim1 Release (L1)
Lim1 Gain (L1)
Lim2 Threshold (L2)
Lim2 Ratio (L2)
Lim2 Attack (L2)
Lim2 Release (L2)
Lim2 Gain (L2)
Lim3 Threshold (L3)
Lim3 Ratio (L3)
Lim3 Attack (L3)
Lim3 Release (L3)
Lim3 Gain (L3)
Lim4 Threshold (L4)
Lim4 Ratio (L4)
Lim4 Attack (L4)
Lim4 Release (L4)
Lim4 Gain (L4)

BOSE®

Better sound through research®

©2002 Bose Corporation, The Mountain
Framingham, MA 01701-9168 USA
www.bose.com
262350 AM Rev.00 JN20793

SPECIFICATIONS SM109z

FEATURES

- Compact 2-way, 10-in stage monitor/PA
- Dual purpose loudspeaker: stage monitor and PA
- Rubber feet included for floor use plus a pole mount cup for stand mounting
- Passive crossover and EQ for economical single amplifier operation
- Dual Neutrik NL4 connectors for input and loop through connection
- Integral handle strategically placed for easy transport

DESCRIPTION

The SM109z is a compact, professional loudspeaker configured for dual-purpose stage monitor and PA use. It is designed for applications with limited space requiring high quality monitoring. The enclosure is fitted with a pole cup for upright use, increasing the SM109z's usefulness as a main PA loudspeaker. The conical HF waveguide provides wide-angle projection, important in smaller venues where a single monitor must often cover several performers. As a PA loudspeaker, this same pattern is ideal for meetings and presentations. Portable usage is enhanced by the heavy-duty enclosure construction, steel grille, and integral handle. Two Neutrik NL4 jacks provide reliable input and loop through connections. The SM109z carries a six year warranty.

APPLICATION

Applications Include:

- Houses of worship
- TV studios
- Meeting rooms
- Club Bands
- Theatres
- Classrooms

PERFORMANCE

Frequency Response (1 W @ 1m)	
±3 dB	100 Hz to 19 kHz
-10 dB	80 Hz
Axial Sensitivity (dB SPL, 1 Watt @ 1m)	
Full Range	95
Impedance (Ohms)	
Full Range	8
Power Handling (Watts)	
Full Range	300
Recommended High-Pass Frequency	
(24 dB/octave)	70 Hz
Calculated Maximum Output (dB SPL)	
Full Range Peak	126
Full Range Long Term	120
Nominal Coverage Angle / -6 dB points (degrees)	
Conical	100

PHYSICAL

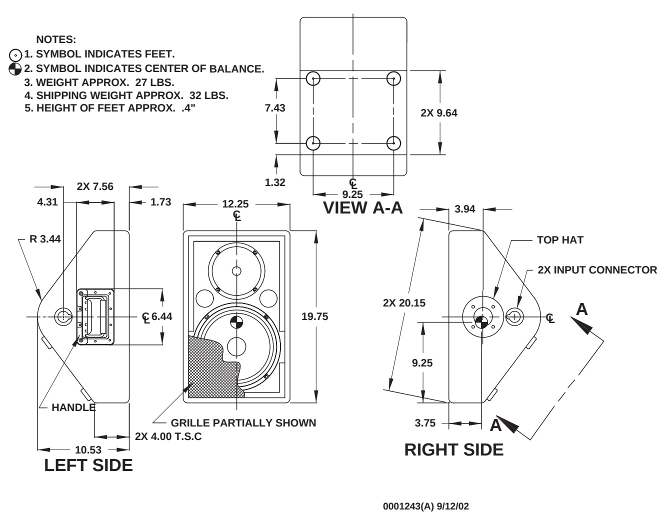
Part Number	0001307	
Configuration	2-way, full-range floor monitor	
LF Subsystem	1x 10-in, vented	
HF Subsystem	1-in exit/1.4-in voice coil compression driver on Wave Guide Plate™	
Powering Mode	Passive LF/HF crossover	
Enclosure Materials	Exterior grade Baltic birch plywood	
Finish	Wear resistant textured black paint	
Connectors	2x Neutrik NL4 Speakon	
Mounting Hardware	Pole mount cup	
Grille	Powder coated perforated steel	
Dimensions	**inches**	**millimeters**
Height	14.88	378
Height w/feet	15.5	394
Width	12.25	311
Depth	18.00	457
Floor Angle	37°	
Weights	**pounds**	**kilograms**
Net Weight	30	13.7
Shipping Weight	33	15.0

EAW products are continually improved. All specifications are therefore subject to change without notice.

SM109z/0004644/001/2 pp September 2002 Printed in USA

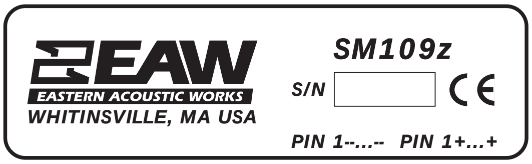
SPECIFICATIONS SM109z

DIMENSIONAL DRAWING


NOTES:
1. SYMBOL INDICATES FEET.
2. SYMBOL INDICATES CENTER OF BALANCE.
3. WEIGHT APPROX. 27 LBS.
4. SHIPPING WEIGHT APPROX. 32 LBS.
5. HEIGHT OF FEET APPROX. .4"
7.43
2X 9.64
1.32
9.25
VIEW A-A
2X 7.56
4.31
1.73
12.25
R 3.44
6.44
19.75
HANDLE
GRILLE PARTIALLY SHOWN
2X 4.00 T.S.C
10.53
LEFT SIDE
3.94
TOP HAT
2X INPUT CONNECTOR
2X 20.15
9.25
3.75
A
A
RIGHT SIDE
0001243(A) 9/12/02

Manufacturing tolerances are +/- 0.13 and +/- 1°

INPUT


EAW
EASTERN ACOUSTIC WORKS
WHITINSVILLE, MA USA
SM109z
S/N
CE
PIN 1-...-- PIN 1+...+

NEUTRIK
NL 4 MPR

A & E SPECIFICATIONS

The 2-way full-range loudspeaker system shall incorporate a 10-in LF transducer and a 1.4-in voice coil compression driver.

The LF transducer shall be mounted in a vented enclosure tuned for optimum low frequency response. The HF transducer shall be loaded on a axis-symmetrical, 1-in exit Wave Guide Plate™ with a nominal conical beamwidth of 100°. An internal passive filter network shall provide all acoustical crossover and system equalization requirements.

System frequency response shall vary no more than ±3 dB from 100 Hz to 19 kHz measured on axis. The loudspeaker shall produce a sound pressure level (SPL) of 95 dB SPL on axis at 1 meter with a power input of 1 watt, and shall be capable of producing a peak output of 126 dB SPL on axis at 1 meter. The loudspeaker shall handle 300 watts of amplifier power (continuous) and shall have a nominal impedance of 8 ohms.

The loudspeaker enclosure shall be irregularly trapezoidal in shape and shall have rubber footpads on one side. Its baffle shall be angled up 37°. It shall be constructed of exterior grade Baltic birch plywood and shall employ extensive internal bracing. It shall be finished in wear-resistant textured black paint. Input connectors shall be 2x Neutrik NL4 Speakon wired in parallel. It shall include a pole mount cup and handle integral to the enclosure. The front of the loudspeaker shall be covered with powder coated perforated steel grille.

The 2-way full-range loudspeaker shall be the EAW model SM109z.

User's Manual

LT-800 Stationary Transmitter

LT-700 Portable Transmitter

LR-500 Programmable Display Receiver

LR-400 Display Receiver

LR-300 Digital Receiver

(Both 72 and 216MHz Systems)

LISTEN

TECHNOLOGIES

CORPORATION

© Copyright 2001 Listen Technologies Corporation

Dear Valued Customer,

Thank you for choosing Listen! All of us at Listen are dedicated to providing you the highest quality products and prompt, efficient customer care. Our products are manufactured in an ISO-9000 factory that has been independently certified to the highest quality standards. We stand ready to answer any questions you might have at installation or in the operation of our products. And should there be any problems with your Listen products, we are ready to help you in any way we can. Should you have any comments on how we might improve our products or our service, we're here to listen. Here's how to reach us:

Telephone: +1.801.233.8992
Facsimile: +1.801.233.8995
U.S. Toll Free: 1.800.330.0891
E-Mail: support@listentech.com
Web: www.listentech.com

Thank you... and enjoy your listening experience!

Best Regards,
The Listen Team

Table of Contents

MISCELLANEOUS

Technical Service Contacts 4
Warnings 4
Compliance Notices 4
Limited Lifetime Warranty 5

LT-800 Stationary Transmitter

LT-800 front illustration: controls & display descriptions .. 7
Audio connections illustration 7
Preparing the LT-800 for use 8
Lock into or unlock a channel 8
Audio control instructions 9
Operating the unit 10
Selecting a channel to transmit on 10
LT-800 specifications 11
LT-800 troubleshooting 12

LT-700 Portable Transmitter

Preparing the LT-700 for use 13
Operating the unit 14
Selecting a channel to transmit on 15
Look & Listen™ LCD display descriptions 15
LT-700 programming 16
Locking into only one channel 16
LED indicator 16
Battery status icon 16
Locking the access door 17
Charging batteries using SmartCharge™ 17
Operating the unit with a wall transformer 18
LT-700 troubleshooting 19
LT-700 specifications 20

LR-500
Programmable
Display
Receiver

LR-400
Display
Receiver

LR-300
Digital
Receiver

LR-500 Programmable Display Receiver, LR-400 Display Receiver and LR-300 Digital Receiver

Differences in the LR- 300, 400 and 500 21
Preparing Listen receivers for use 21
Operating the LR-300 receiver 22
Operating the LR-400 and LR-500 receivers 22
Locking into only one channel 23
LED indicator 23
LR-400 and LR-500 Look & Listen™
LCD display descriptions 23
Battery status icon 24
Unlocking and locking the access door 24
Charging batteries using SmartCharge™ 24
Operating the receiver with a wall transformer 25
LR-300, LR-400 and LR-500 specifications 26
Receiver troubleshooting 27

LR-500 programming 25

Frequency & Compatibility Tables

72MHz 28-29
216MHz 30-31

Technical Service Contacts

If technical service is needed, please contact Listen. Pre-authorization is required before returning Listen products. If products were damaged in shipment, please contact the carrier, then contact Listen for replacement or repair requirements payable by the carrier.

Toll Free: 1.800.330.0891
Telephone: +1.801.233.8992
Fax: +1.801.233.8995
Internet: www.listentech.com
General E-Mail: info@listentech.com

Shipping & Correspondence Address:
Listen Technologies Corporation
8535 South 700 West, Suite A
Sandy, Utah 84070 U.S.A.

FCC ID Numbers:

LT-800-072 (72MHz base transmitter) ID: OMD800-001
LT-800-216 (216MHz base transmitter) ID: OMD800-216
LT-700-072 (72MHz portable transmitter) ID: OMD700-001
LT-700-216 (216MHz portable transmitter) ID: OMD700-216

Compliance Notices

Listen's Models LT-800, LT-700, LR-300, LR-400 and LR-500

These devices comply with part 15 of the FCC Rules. Operation is subject to the following two conditions: (1) These devices may not cause harmful interference, and (2) these devices must accept any interference received, including interference that may cause undesirable operation.

Listen's LT-800 and LT-700 Transmitters (216MHz only)

Listen's LT-800 and LT-700 transmitters are authorized by rule under the Low Power Radio Service (47 C.F.R. Part 95) and must not cause harmful interference to TV reception or United States Navy SPASUR installations. You do not need an FCC license to operate these transmitters. These transmitters may only be used to provide: auditory assistance to persons with disabilities, persons who require language translation, or persons in educational settings; health care services to the ill; law enforcement tracking services under agreement with a law enforcement agency; or automated maritime telecommunications system (AMTS) network control communications. Two-way voice communications and all other types of uses not mentioned above are expressly prohibited.

Caution: Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate Listen's equipment.

FCC Statement

This equipment has been tested and found to comply with the limits for a class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

This equipment has been certified to comply with the limits for a class B computing device, pursuant to FCC Rules. In order to maintain compliance with FCC regulations, shielded cables must be used with this equipment. Operation with non-approved equipment or unshielded cables is likely to result in interference to radio and TV reception. The user is cautioned that changes and modifications made to the equipment without the approval of manufacturer could void the user's authority to operate this equipment.

Limited Lifetime Warranty, Including Performance Guarantee

Listen Technologies Corporation (Listen) warrants its products to be free from defects in workmanship and material under normal use and conditions for the useful lifetime of the product from date of purchase. Accessories are warranted for ninety days from date of purchase. This warranty is only available to the original end purchaser of the product and cannot be transferred. Warranty is only valid if warranty card has been returned within 90 days of purchase. This warranty is void if damage occurred because of misuse or if the product has been repaired or modified by anyone other than a factory authorized service technician. Warranty does not cover normal wear and tear on the product or any other physical damage unless the damage was the result of a manufacturing defect. Listen is not liable for consequential damages due to any failure of equipment to perform as intended. Listen shall bear no responsibility or obligation with respect to the manner of use of any equipment sold by it. Listen specifically disclaims and negates any warranty of merchantability or fitness of use of such equipment including, without limitation, any warranty that the use of such equipment for any purpose will comply with applicable laws and regulations. The terms of the warranty are governed by the laws of the state of Utah.

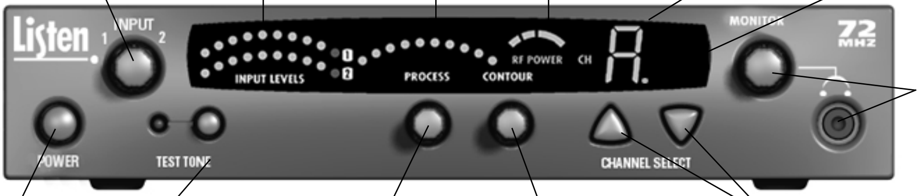
Listen will only accept returned products with prepaid shipping and with a return authorization number. To receive a return authorization number call **1.800.330.0891** or +**1.801.233.8992**.

Listen will refund the purchase price of the product to the original purchaser within the first ninety days after purchase if the product does not perform better than a similar competitive product subject to an annual amount to any one original purchaser not to exceed $1,000 and subject to the conditions of this limited warranty. This is called our Whatever-It-takes Performance Guarantee. Refunds will only be given to the original purchaser of product once the product has been returned to the factory in good working order with an explanation for the refund. To receive a refund, call Listen for a return authorization number. Return the product with a letter on the end user's letterhead stating the reason for the refund and a copy of the original invoice. If the product was purchased through a dealer, refunds must be processed through the original dealer.

In the first ninety days after purchase, any defective product will be replaced with a new unit. After ninety days, Listen will, at its own discretion either replace the unit with a new unit or a unit of similar type and condition. Product that is not covered under warranty shall be replaced with a unit of similar type and condition based on a flat fee.

This limited lifetime warranty, prices and the specifications of products are subject to change without notice.

LT-800 Front Illustration: Controls & Displays



INPUT: Adjust audio input levels of Inputs 1 and 2

Indicates audio input level of Inputs 1 and 2

Shows modulation level after processing

Indicates RF output power (full, 1/2 or 1/4) – select on rear panel

Channel the unit is tuned to

If decimal place is lit, channel is locked

Plug in a headset then adjust volume

Power ON/OFF

Activates a tone to aid tuning of non-Listen receivers

PROCESS: Adjust the amount of audio processing – to make all audio sources sound like they are at the same level, adjust clockwise

CONTOUR: Adjust voice intelligibility equalization for your application requirements

SELECT CHANNEL

TO LOCK: press both UP and DOWN buttons simultaneously to lock channel; press both again to unlock

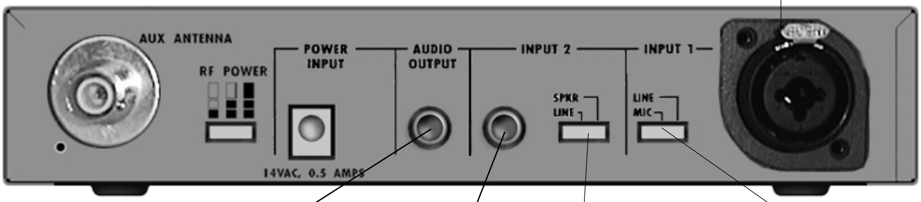
Please Note: as of December, 2002, "Process" has been relabeled "MIX"

LT-800 Audio Connections

The LT-800 has two possible audio inputs:

INPUT 1 is used to connect a balanced mic or line level input.

INPUT 2 is used to connect an unbalanced, high impedance line input or speaker input.



INPUT 1 CONNECTION: Connect a microphone or balanced line level input. See Preparation instruction #7 for details (Page 9).

LINE OUTPUT – the LINE OUTPUT connector is a mixed output of INPUT 1 and INPUT 2 after processing. This is an unbalanced phono connector. The output level is –10dBm.

INPUT 2 CONNECTION: this is a phono connector used to connect an unbalanced, high impedance line or speaker input

INPUT 2 SWITCH: Select LINE or SPEAKER on the slide switch directly next to the INPUT 2 connector. In the line position, the input level is –10dBm, and on speaker, the input level is +10dBm.

INPUT 1 SWITCH: Select MIC or LINE input on the slide switch. For Mic input, the level is –55dBm and the input impedance is 50 ohms. In the line position, the input level is +4dBm and the input impedance is 600 ohms.

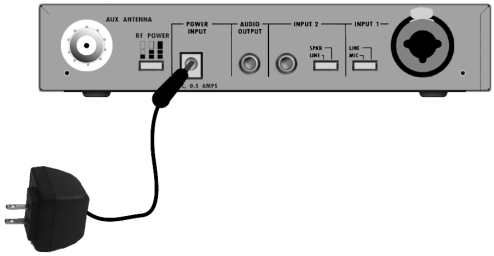
Listen™

Preparing the LT-800 for Use

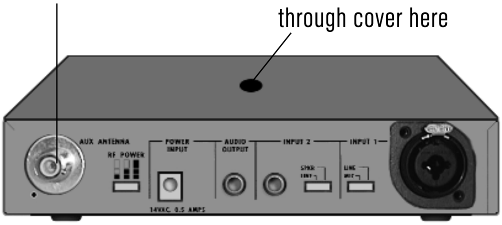
1 **Remove the product from the shipping container and plastic cover.** Inspect for physical damage. See "Technical Service Contacts" (Page 4) if damage has occurred.

2 **For Rack Mounting:** If rack mounting the unit, install the rack mount shelf according the instructions that come with the shelf, then install the unit and shelf in the rack.

3 **Plug in Wall Transformer** Plug the connector on the wall transformer into the power connector on the back panel. Do not plug into power source yet. (Only use a Listen supplied transformer.)


AUX ANTENNA
RF POWER
POWER INPUT
AUDIO OUTPUT
INPUT 2
INPUT 1
SPKR LINE
LINE MIC
14VAC, 0.5 AMPS

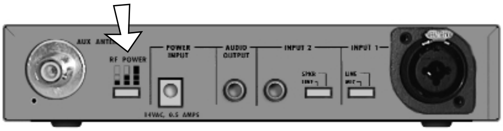
4 **Connect Antenna:** Connect the supplied antenna according the instructions that come with the antenna. Only use an antenna supplied by Listen.


AUX ANTENNA
RF POWER
POWER INPUT
AUDIO OUTPUT
INPUT 2
INPUT 1
14VAC, 0.5 AMPS

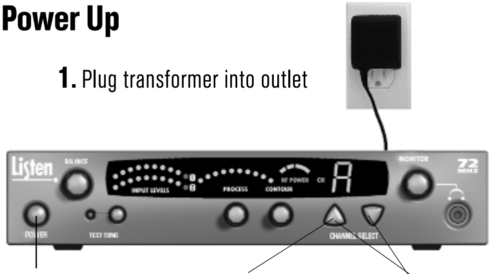
Connect external antenna here, **or...**

Connect telescoping or rubber ducky antenna through cover here

5 **Set the RF POWER:** Set the RF POWER switch for FULL, 1/2 or 1/4. (Level is also indicated on the front panel.) The amount of transmitted RF power that you will need depends on your application. If you are operating other transmitters in the same environment and receivers can be used close to the antenna, it is best to have the transmitter's output power at its lowest level to reduce the possibility of interference.


AUX ANTENNA
RF POWER
POWER INPUT
AUDIO OUTPUT
INPUT 2
INPUT 1
14VAC, 0.5 AMPS

6 **Power Up**


Listen
BALANCE
INPUT LEVELS
PROCESS
CONTOUR
RF POWER
CH
MONITOR
72
POWER
TEST TONE
CHANNEL SELECT

1. Plug transformer into outlet

2. Press to turn power ON

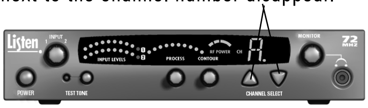
3. Select channel using UP and DOWN buttons

4. If desired, lock in selected channel by pressing both UP and DOWN buttons simultaneously

Lock Into or Unlock A Channel

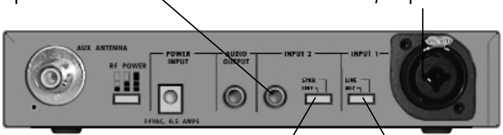
TO LOCK: Press the LT-800's channel UP and DOWN buttons simultaneously for three or more seconds until you see a decimal dot light up next to the channel number.

To UNLOCK: Press the channel UP and DOWN buttons simultaneously for three or more seconds until you see the decimal dot next to the channel number disappear.


Listen
INPUT
INPUT LEVELS
PROCESS
CONTOUR
RF POWER
CH
MONITOR
72
POWER
TEST TONE
CHANNEL SELECT

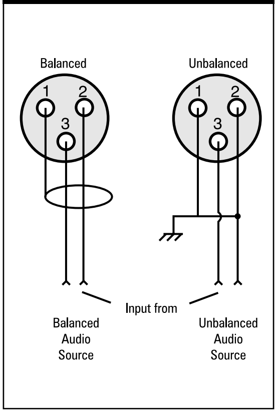
Listen™

7 **Connect Audio Inputs:** Connect audio source(s) to one or both audio input connections: Input 1 offers a choice of balanced XLR or 1/4" phone connection. Input 2 is an unbalanced phono connector. (See LT-800 Audio Connections on Page 7 for specific connection instructions.)

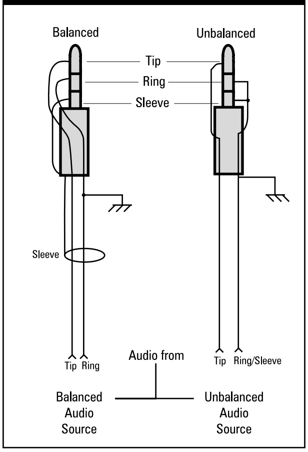

Unbalanced Input 2 connection: phono connector
Balanced Input 1 connection: XLR or 1/4" phone
AUX ANTENNA
RF POWER
POWER INPUT
AUDIO OUTPUT
INPUT 2
INPUT 1
SPKR LINE
LINE MIC
14VAC, 0.5 AMPS
Input 2 Switch
Input 1 Switch

Connect balanced and unbalanced audio sources as shown in diagram. Make sure that if you are connecting an unbalanced audio source, that Pin 2 of the XLR and the ring of 1/4" phone jack are connected to ground as shown.

XLR Wiring


Balanced
Unbalanced
1
2
3
1
2
3
Input from
Balanced Audio Source
Unbalanced Audio Source

1/4" Phone Wiring


Balanced
Unbalanced
Tip
Ring
Sleeve
Sleeve
Tip
Ring
Audio from
Tip
Ring/Sleeve
Balanced Audio Source
Unbalanced Audio Source

8 **Adjust Audio Input Switches** – For Input 1 Switch, select MIC or LINE input level; Input 2 Switch, select LINE or speaker level. (See illustration on Page 7.)

9 **Use the headset jack to listen to audio source** – With the audio source or sources active and the unit powered ON, plug a headset into the jack located on the LT-800's front panel to listen to the audio source. Adjust headset volume to a comfortable level.

LT-800 Audio Control

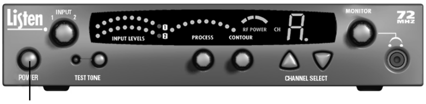
(Refer to LT-800 Controls & Displays Illustration on Page 7.)

The LT-800 incorporates automatic audio level control and processing. Input levels can be viewed on the two VU meters on the front panel.

1. **ADJUST INPUT:** Adjust INPUT knob counter-clockwise to add gain to INPUT 1. Adjust INPUT knob clockwise to add gain to INPUT 2. If you have two audio sources connected to both INPUT 1 and 2, adjust the level of one input using the VU meter, then adjust the output level of the other audio source.
2. **ADJUST CONTOUR:** Adjust the CONTOUR knob counter-clockwise. If most of your audio source is voice, adjust clockwise for music. The CONTOUR knob adjusts the relative equalization of the unit. This equalization boosts or cuts frequencies above 3kHz.
3. **ADJUST PROCESS:** Adjust the PROCESS knob clockwise if you want more processing. This will provide for even more consistent levels. This knob adjusts processing, relative to automatic processing already performed in the unit. The LT-800 provides dynamic audio processing intended to keep audio levels consistent to the listener. When the PROCESS knob is rotated fully clockwise, all audio is at the same relative level. When rotated fully counter clockwise, more dynamic range is provided for the transmitted audio source.

Operating the LT-800

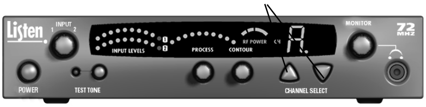
1 Make sure the unit is powered on.



Press POWER button and verify power connections. When “ON”, display should be lit.

2 Select a channel

– Select the channel to transmit on to by pressing the channel UP and DOWN buttons. See "Selecting a Channel to Transmit On" to the right.



Can’t change the channel? It may be locked. Press both the UP and DOWN buttons simultaneously for 3 seconds until the decimal dot next to the channel turns off. Press again to lock in new channel, if desired.

At 72MHz: The LT-800 at 72MHz operates on 17 wide band channels and 40 narrow band channels:

Letters = Wide Band Channels
Numbers = Narrow Band Channels

At 216MHz: The LT-800 at 216MHz operates on 19 wide band channels and 38 narrow band channels:

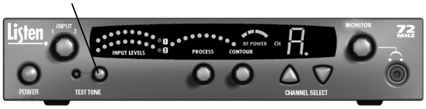
“2” as left digit = wide band channel
“1” and “3” left digits = narrow band channels

Please refer to the Frequency Compatibility Tables (Pages 28-31) for specific frequencies and compatibility with other manufacturers.

- **NOTE:** Operating on a narrow band channel will have more noise.

3 Test Tone, if necessary:

To broadcast a tone, press the test tone button. This button is used to tune other manufacturer receivers or to identify a particular channel on a receiver.



Selecting a Channel to Transmit On

Selecting a clear channel to broadcast is important for interference-free broadcasting.

Wide Band Recommendation – If you don’t plan to have other transmitters (or a limited number) operating in the same area, it is recommended that you use wide band channels. These are the most common channels used. Many people own receivers that will operate on these channels. In addition, wide band channels offer the best fidelity and the lowest noise. In this case, select one of the 17 wide band channels A through R on 72MHz systems (19 wide band channels beginning with “2” on 216MHz systems). The best way to select a channel is by trial and error. Try a channel, if it has low interference and reliable performance, stick to that channel.

Compatibility with other manufacturers – If you are using other manufacturer’s receivers with the LT-800, find out their frequency, then refer to Listen’s Frequency Compatibility Tables (Pages 28-31) to find the LT-800 channel number or letter that corresponds with the receiver’s frequency. As much as possible, Listen has duplicated the wide band channel letters to cover frequencies of other major brands. However, we recommend verifying corresponding channel numbers and letters on the tables on pages 28-31.

If you are using other manufacturer’s receivers and the sound is distorted*, it is probably because the receiver is not designed to handle the ±25kHz deviation of the Listen transmitter. This can be corrected by turning the PROCESS control knob all the way counter-clockwise and turning the INPUT LEVEL knob down.*

Several transmitters operating in the same environment – For this, you’ll need to choose your transmitting frequencies carefully. The following guidelines should be used to choose channels:

a. If possible, use all wide band channels. These channels offer lower noise.

b. Select channels that have the highest frequency separation as possible (see Frequency Compatibility Tables, pages 28-31), to avoid adjacent channels and interference. Channels should be at least 200kHz apart.

c. Through a process of trial and error select a combination of channels that offer the lowest interference and highest reliability.

LT-800 Stationary Transmitter Specifications

RF Frequency Range:
72MHz: 72.05 - 75.95MHz
216MHz: 216.025 - 216.975 MHz

Transmitter Stability:
50 PPM

Output Power:
72MHz: 8000uV at 3 meters
216MHz: 100mW
(Max allowed by FCC)

Signal to Noise ratio:
72MHz: Wide band channels, 60dB;
Narrow, 54dB
216MHz: Wide band channels, 50dB;
Narrow, 44dB

Antenna:
72MHz: Telescoping, Rubber Ducky, Front Mount, Remote Base, or Coaxial Dipole
216MHz: Telescoping, Whip, Front Mount, Remote Base, Ground Plane, or Coaxial Dipole

Available Channels:
72MHz: 17 wide band, 40 narrow band
216MHz: 19 wide band, 38 narrow band
(Modulation is automatically switched when changing between narrow and wide band channels)

Frequency Response:
72 50Hz to 17kHz, ±3dB
216 50Hz to 8kHz, ±3dB

RF Power Selection:
Full, 1/2, 1/4

Physical Dimensions:
8" W x 8"D x 1.75"H
(20.3cm W x 20.3cm D x 4.45cm H)
Optional rack mount brackets

Weight:
3 lbs. (1.4 kg)

Power:
16 VAC/850mA

Audio Input 1:
F-XLR / 1/4" Phone, 50/600 ohms, balanced
-55dBm / +4dBm, selectable, adjustable

Audio Input 2:
Phono, 10k/50 ohms, unbalanced
-10dBm / +20dBm, selectable, adjustable

Audio Output (Mix):
Phono, 10k ohms, unbalanced, -10dBm

Headphone Output:
250mW, 32 ohms, 3.5MM stereo

Controls:
Mix, Process, Contour, Headphone Level, Tone, RF Power Output, and ON/OFF

Visual Indicators:
Audio input levels 1, 2 and post processed modulation level; channel; RF power, test tone

Specifications subject to change without notice.

LT-800 Stationary Transmitter Specifications

RF Frequency Range:
72MHz: 72.05 - 75.95MHz
216MHz: 216.025 - 216.975 MHz

Transmitter Stability:
50 PPM

Output Power:
72MHz: 8000uV at 3 meters
216MHz: 100mW
(Max allowed by FCC)

Signal to Noise ratio:
72MHz: Wide band channels, 60dB;
Narrow, 54dB
216MHz: Wide band channels, 50dB;
Narrow, 44dB

Antenna:
72MHz: Telescoping, Rubber Ducky, Front Mount, Remote Base, or Coaxial Dipole
216MHz: Telescoping, Whip, Front Mount, Remote Base, Ground Plane, or Coaxial Dipole

Available Channels:
72MHz: 17 wide band, 40 narrow band
216MHz: 19 wide band, 38 narrow band
(Modulation is automatically switched when changing between narrow and wide band channels)

Frequency Response:
72 50Hz to 17kHz, ±3dB
216 50Hz to 8kHz, ±3dB

RF Power Selection:
Full, 1/2, 1/4

Physical Dimensions:
8" W x 8"D x 1.75"H
(20.3cm W x 20.3cm D x 4.45cm H)
Optional rack mount brackets

Weight:
3 lbs. (1.4 kg)

Power:
16 VAC/850mA

Audio Input 1:
F-XLR / 1/4" Phone, 50/600 ohms, balanced
-55dBm / +4dBm, selectable, adjustable

Audio Input 2:
Phono, 10k/50 ohms, unbalanced
-10dBm / +20dBm, selectable, adjustable

Audio Output (Mix):
Phono, 10k ohms, unbalanced, -10dBm

Headphone Output:
250mW, 32 ohms, 3.5MM stereo

Controls:
Mix, Process, Contour, Headphone Level, Tone, RF Power Output, and ON/OFF

Visual Indicators:
Audio input levels 1, 2 and post processed modulation level; channel; RF power, test tone

Specifications subject to change without notice.

LT-800 Troubleshooting

The LT-800 has no power.

Make sure the power transformer is connected to a power source and is connected to the jack marked "Power Input". Make sure the POWER button is pressed.

There is no audio or the audio level is too low.

Make sure that you have an audio source properly connected to INPUT 1 and/or 2. The INPUT 1 or 2 switch must be located in the correct position for the appropriate input level. For example: if you are using the output of a mixer on INPUT 2, the switch should be in the LINE position. If it were in the SPEAKER position, the level would be too low. Also, check the INPUT knob to ensure it is properly adjusted. You should be able to see the VU meter deflect on INPUTS 1 or 2 corresponding with the input level of the audio source. Try connecting a headset to the front panel jack and adjusting the MONITOR volume control so that you can listen to the audio source.

The audio is distorted.

Check to make sure you have the input level select switches in the proper position. You may be providing too much audio level for the input stage to handle.

There is hum in the audio.

Make sure you have properly grounded the audio source to the LT-800. Check the connections from the audio source to the LT-800. If you can, try to use a balanced audio source - this will reduce the chance of creating hum. Try connecting a ground from the LT-800 to ground and/or to the ground of the source audio.

There is too much audio processing.

Adjust the process knob counter-clockwise.

Please note: PROCESS has been relabeled MIX

There is a tone.

The TEST TONE button has been pressed (its LED light is on). Push the TEST TONE button to turn off the tone.

I cannot change the broadcast channel.

The unit is locked. To unlock, press the UP and DOWN button simultaneously for 3+ seconds. You can tell when it's locked because the decimal is illuminated.

The Audio Input 1 sounds “tinny”.

If you are using an unbalanced audio source, make sure Pin 2 or the XLR and the ring on the 1/4” jack is grounded.

There is interference.

Try different frequencies until you find a clear channel. If this does not work, try a different frequency band (i.e. if you're using 72MHz, try 216MHz or vice versa). This is accomplished by returning the equipment to Listen (no charge) and replacing it with Listen’s alternate frequency band equipment.

I cannot pick up the signal on the receiver.

Check to make sure the receiver and the transmitter are on the same frequency.

I can pick up the signal on the receiver, but it sounds like it's not tuned in.

Check to make sure the transmitter and receiver are on exactly the same channel number/letter.

I'm using another brand of receiver - how do I tell which channel to use?

Refer to Listen’s Frequency Compatibility Tables (pages 28-31). Adjust Listen’s transmitter or receiver to the same frequency as the other major brand. Since Listen products can access 57 channels, they will most likely transmit or receive on the same fixed channel or channels of other major ALD brands.

There is not sufficient range.

First make sure that the receivers you are using are operating properly. Then, make sure that you have an antenna connected either to the top of the LT-800 transmitter or connected to the back of the unit (but not both!). Locate the antenna at a location as high as possible and free of obstacles. Make sure the antenna is of the correct type and is connected properly to the unit. Try using different frequencies that don't have interference. If you're using 216MHz, you might want to try a directional antenna.

End users are adjusting the unit.

First, lock the channel by pressing the UP and DOWN buttons together for 3+ seconds. Consider removing the INPUT, PROCESS AND CONTOUR knobs. You can order from Listen a rack mount kit and security cover that will limit access to unit from end users.

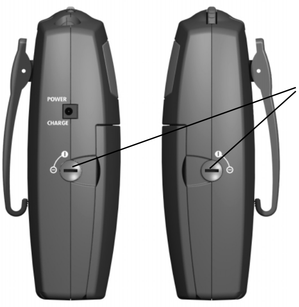
Other manufacturers’ receivers sound distorted.

It is probably because the Listen transmitter is over-deviated. See Page 10 for instructions on how to correct this.

Preparing the LT-700 for Use

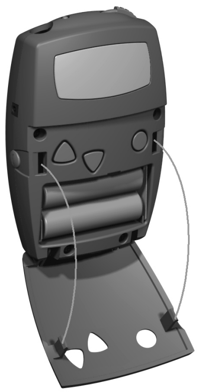
1 **Remove the product from the shipping container** and plastic cover. Inspect for physical damage. See "Technical Service Contacts" (Page 4) if damage has occurred.

2 **Check to see if the door is locked.** If locked, use a pocketknife or small screwdriver to unlock the door locks on the both sides of the unit. To unlock the door, rotate the lock 1/4 turn counter clockwise.



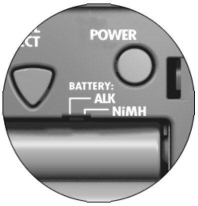
On both the right and left side, unlock by turning slot horizontal as shown.

3 **Open the front access door** by gripping the two tabs with your thumb and index finger.



4 **Select Battery Type** – **CAUTION:** If you are using any battery type other than rechargeable Nickel Metal Hydride (NiMH) batteries, make sure the BATTERY SELECT switch is in the **alkaline** position (located in the battery compartment and labeled BATTERY SELECT, remove batteries to access this switch). The product is shipped from the factory in the alkaline position.

To use NiMH batteries, before placing in unit, change the battery switch to NiMH. (See page 17 for more details.)

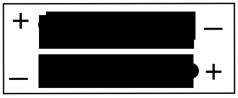


BATTERY SELECT Switch - With batteries removed, use a pen or small screwdriver to select the battery type

WARNING: Do not place the BATTERY SELECT switch in the NiMH position when using Non-Nickel Metal Hydride batteries. The NiMH position will attempt to charge the batteries. Charging non-Nickel Metal Hydride batteries may result in physical harm, destruction of property and/or fire.

5 **Place Batteries in Unit** Place two AA batteries in the compartment, making note of the battery polarity shown in the battery compartment, and again verifying that the BATTERY SELECT switch is switched correctly for your batteries (see Step 4).

Top of Unit and Battery Compartment

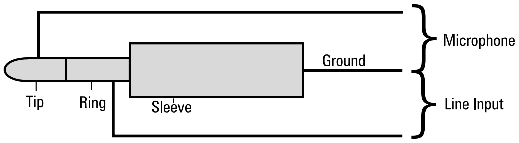


LT-700 Portable Transmitter Preparation (cont.), Operation

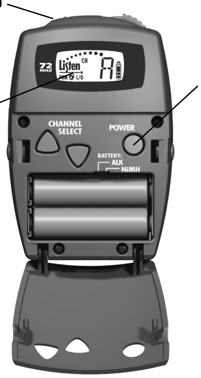
Preparing the LT-700 for Use (continued)

6 **Connect the microphone** to the microphone jack on the top of the unit. The unit uses the microphone cable as an antenna. If you are using the line input, use the special cable accessory designed for this purpose.

Or, connect optional Line Level Input Cable – Use an optional Listen supplied line input cable (LA-263), or connect as follows:



7 **Turn the unit on.** If it does not turn on by pressing power button, check to ensure you have installed charged batteries and that the batteries have been installed with the proper polarity.



2. LED should light up red when ON

3. LCD display should show battery level, channel and the Listen logo

1. Turn POWER on by pressing button, turn off by pressing again

Operating the LT-700 Transmitter

1 **Make sure the unit is on** by opening the access door and pressing the POWER button.

2 **Select the channel** to transmit on by pressing and holding the channel UP and DOWN buttons for 3 seconds. The LOCK symbol on the LCD will go off and the channel can be changed. Also see “Selecting a Channel to Transmit On”.

Operating the LT-700: Select a Channel (continued)

- **At 72MHz,** the LT-700 operates on 17 wide band channels and 40 narrow band channels (letters = wide band and numbers = narrow band channels).
- **At 216MHz,** the LT-700 operates on 19 wide band channels and 38 narrow band channels (channels beginning with “2” = wide band, and channels beginning with “1” or “3” = narrow band channels).

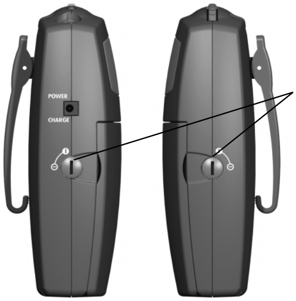
Please refer to the Frequency Compatibility Tables (Pages 28-31) for specific frequencies and compatibility with other manufacturers.

- **NOTE: Operating on a narrow band channel will have more noise.**

Close the access door.

If you choose, lock the door by rotating the locks on the right side 1/4 turn clockwise, and on the left side 1/4 turn counter clockwise. This can be done with a small screwdriver or pocketknife.

Access Door Closed



On both the right and left side, lock door by turning slot vertical as shown.

Selecting a Channel Transmit On

Selecting a clear channel to broadcast is important for interference free broadcasting.

Wide Band Recommendation: If you don't plan to have other transmitters (or a limited number) operating in the same area, it is recommended that you use wide band channels. These are the most common channels used. Many people own receivers that will operate on these channels. In addition, wide band channels offer the best fidelity and the lowest noise. In this case, select one of the 17 wide band channels A through R on the 72MHz system, or one of the 19 wide band channels on the 216MHz system labeled 2A, 2B, 2C, etc. The best way to select a channel is by trial and error. Try a channel, if it has low interference and reliable performance, stick to that channel.

Compatibility with other manufacturers – If you are using other manufacturer's receivers with the LT-700, find out their frequency, then refer to Listen's Frequency Compatibility Tables (Pages 28-31) to find the LT-700 channel number or letter that corresponds with the receiver's frequency. As much as possible, Listen has duplicated the wide band channel letters to cover frequencies of other major brands. However, we recommend verifying corresponding channel numbers and letters on the tables on pages 28-31.

If you are using other manufacturer's receivers and the sound is distorted*, it is probably because the receiver is not designed to handle the ±25kHz deviation of the Listen transmitter.*

Several transmitters operating in the same environment – For this, you will need to choose your transmitting frequencies carefully. The following guidelines should be used to choose channels:

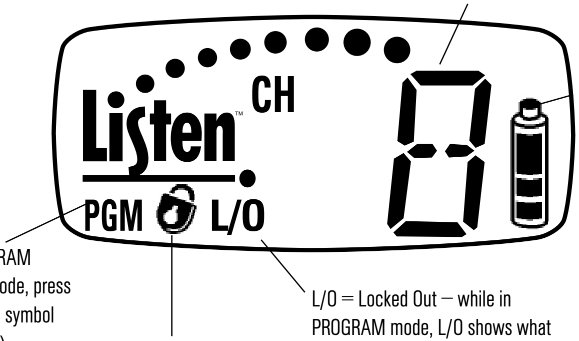
a. If possible, use all wide band channels. These channels offer lower noise.

b. Select channels that have the highest frequency separation as possible (see Frequency Compatibility Tables, pages 28-31), to avoid adjacent channels and interference. Selected channels should be at least 200kHz apart.

c. Through a process of trial and error select a combination of channels that offer the lowest interference and highest reliability.

LT-700 Look & Listen™ Display Descriptions

Channel the unit is tuned to



Indicates battery level in 3 stages, flashes when battery is low along with flashing red LED

Indicates the unit is in PROGRAM mode (to enter PROGRAM mode, press UP and DOWN until the PGM symbol on the LCD display is shown.)

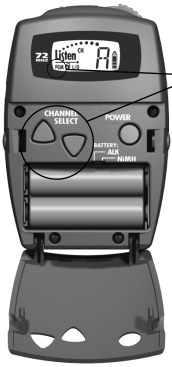
Indicates unit is locked into channel, press and hold UP or DOWN button for 3 seconds to unlock

L/O = Locked Out – while in PROGRAM mode, L/O shows what channels are locked in or out

LT-700 Programming

In General: The LT-700 can be programmed to transmit on a limited number of channels. For applications where users are required to select a channel (such as classrooms or language interpretation), and you don't want them to have to scroll through all of the available channels, this feature is ideal. If all channels where available, the user could have 57 channels to select from.

1. Enter PROGRAM Mode:

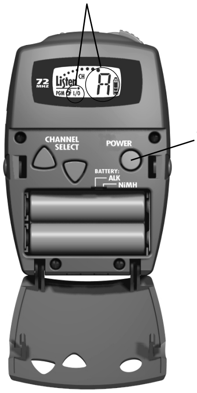


To enter PROGRAM mode, open the access door and press the UP and DOWN keys simultaneously until the PGM symbol is displayed on the LCD display.

2. Scroll through channels to lock or unlock

Use the UP and DOWN channel select keys to scroll through all 57 channels. When checking a particular channel, the L/O symbol on the LCD will be displayed when that channel is locked out. "Locked Out" means the channel is not available to the user after the programmer exits PROGRAM mode.

On each channel, check to see if "L/O" is displayed; it means channel is locked out from the user



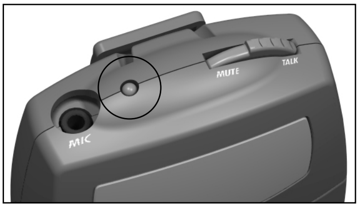
To toggle between locked out and not locked out, press the POWER button.

3. To exit PROGRAM mode, don't press a key for 4 seconds.

Locking into One Channel

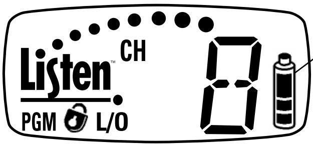
Enter PROGRAM Mode (as described to the right), lock all channels except the channel you want to use.

LED Indicator

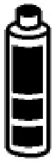


Steady Red LED = Normal Operation
Slow Flashing = Battery is Low
Slow Flashing while charging = Unit is Charging
Fast Flashing = Mute

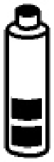
Battery Status Icon



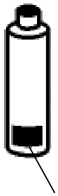
The status of the battery is indicated by the battery symbol on the LCD.



When the battery is fully charged, all three segments of the battery symbol will be displayed, as shown.



As fewer battery segments are displayed, the lower the battery level, as shown.

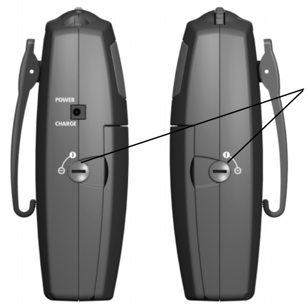


When the batteries are almost fully discharged, the battery segments and the red LED near the volume control will blink. When this occurs, less than 5% of the battery life is remaining and the batteries should be immediately replaced or recharged.

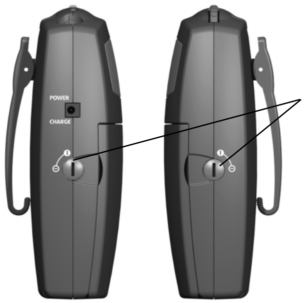
This segment will blink when batteries are low.

Unlocking and Locking the Access Door

The front access door can be locked so that the users will not have access to controls or batteries. To do this, use a pocketknife or small screwdriver to rotate the locks as shown.



To unlock: On both the right and left side, unlock by turning slot horizontal as shown.



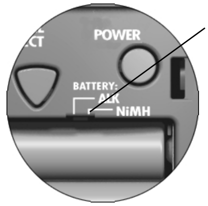
To lock: On both the right and left side, lock door by turning slot vertical as shown.

Charging Batteries Using SmartCharge™

1. The LT-700 is unique because it has an automatic battery charger built in. That means that when the unit is connected to a wall transformer or placed in a Listen battery recharging tray, NiMH batteries will be charged within the LT-700. Once batteries are charged the internal battery charger will automatically turn off. This prevents overcharging of batteries, automates the process of charging, and makes it so you don't have to remember to unplug the charger.
2. Do not attempt to charge alkaline or any other type of non-NiMH batteries. Only charge Nickel Metal Hydride (NiMH) batteries within the LT-700. We recommend that you use NiMH batteries supplied by Listen.

WARNING: Attempting to charge non-NiMH batteries may result in physical harm, destruction of property and /or fire.

3. **IMPORTANT:** To charge NiMH batteries, the BATTERY SELECT switch must be selected to NiMH. Use a pen or small screwdriver to the move the switch (located in the battery compartment directly below the BATTERY SELECT label) to the proper position.



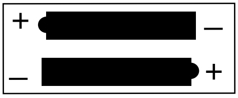
For only NiMH batteries: set BATTERY SELECT switch to NiMH. For all other batteries, set BATTERY SELECT at alkaline.

(Continued...)

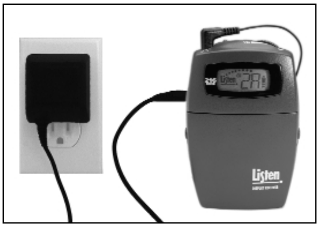
Charging the Batteries (continued)

4. **Install NiMH rechargeable batteries** in the battery compartment, ensuring that proper polarity has been observed.

Top of Unit and Battery Compartment



Note: It is recommended that you only use a Listen charging transformer. Or, set the LT-700 with NiMH batteries within unit and powered ON in a Listen Carrying/Charging Case (LA-311 or LA-317).



For charging using a transformer: with NiMH batteries in unit, plug wall transformer in CHARGING connection and into wall outlet (as shown). Batteries will be fully charged when red LED stops flashing and is unlit. (LED and battery symbol will flash slowly while charging.)

5. **The LT-700 must be powered ON initially to charge NiMH batteries within unit** – Connect a wall transformer to the LT-700 or place unit in a Listen Charging Case (LA-311 or LA-317). The unit must be on to charge. However, once the charger has started, the unit can be turned off.

6. Once the batteries are fully charged, the unit will stop charging the batteries. Full-charge will be indicated when all the battery symbol segments are illuminated and the LED near the volume control stops flashing and is unlit.

7. The unit can be operated while batteries are charging.

Operating the LT-700 with a Wall Transformer

1. The unit will operate normally when connected to a wall transformer.

2. Batteries do not need to be installed in the unit to operate using a wall transformer.

LT-700 Troubleshooting

The LT-700 has no power.

Make sure the unit has either fully charged batteries or a Listen approved wall transformer connected to the unit. Open the door and press the ON button. If this does not work, try a different set of batteries. Make sure the batteries are installed properly.

There is no audio.

Make sure the MUTE switch is in the TALK position. Make sure you have the microphone plugged all the way in the top of unit. Make sure you are using a Listen approved microphone. If you're using the line input, make sure you have connected a line level, unbalanced input to the "ring" of the connector.

The audio is distorted.

Make sure you're using an approved Listen microphone. Make sure the line input level is not too high.

There is hum in the audio.

The microphone may be too close to a transformer.

The microphone is low in level.

The microphone must be placed in close proximity to the person speaking. If this does not work, try using a head-worn microphone.

There is too much noise.

This is probably due to the fact that the microphone is not close enough to the person speaking and is picking up background noise. Try positioning the microphone closer or try using a microphone that is directional (like a head-worn mic).

There is interference.

Try different frequencies until you find a clear channel. If this does not work, try a different frequency band (i.e. if you're using 72MHz, try 216MHz or vice versa). This is accomplished by returning the equipment to Listen (no charge) and replaced with alternate frequency band equipment.

I cannot pick up the signal on the receiver.

Check to make sure the receiver and the transmitter are on the same frequency.

I can pick up the signal on the receiver, but it sounds like it's not tuned in.

Check to make sure the transmitter and receiver are on exactly the same channel number/letter.

I'm using another brand of receiver - how do I tell which channel to use?

Refer to Listen's Frequency Compatibility Table (included on Pages 28-31). Either adjust the transmitter or the receiver to a common channel.

There is not sufficient range.

The LT-700 is a portable transmitter that uses the microphone cable as an antenna and as such, the range will vary depending on the location of the receivers compared to the transmitter. You can only expect 100 feet of effective working area.

It's confusing for users to have 57 channels when switching between channels.

Use the programming feature of the LT-700 to limit the number of channels users have access to.

How can I limit access to the batteries?

Use the side door locks.

My batteries are not charging.

Make sure the battery switch is in the NiMH position. Make sure you have turned the unit's POWER on initially to charge. Make sure you have connected an approved Listen charging transformer. Or if inserted in charging tray, make sure the unit is ON and is pushed securely into tray. Make sure you use ONLY NiMH batteries provided by Listen. Never try to charge alkaline batteries.

I want to run the unit from a wall transformer.

Simply plug a Listen approved transformer to the Charge/Power connectors. Batteries need not be installed, however NiMH will be recharged if installed, and if the BATTERY SELECT switch is set to NiMH.

Listen™

LT-700 Portable Transmitter Specifications

RF Frequency Range:
72MHz:72.05 - 75.95MHz
216MHz:216.025 - 216.975 MHz

Transmitter Stability:
50 PPM

Output Power:
72MHz: 8000uV at 3 meters
216MHz: 100mW
(Max allowed by FCC)

Signal to Noise ratio:
72MHz: Wide band channels, 60dB;
Narrow, 54dB
216MHz: Wide band channels, 50dB;
Narrow, 44dB

Antenna:
Uses microphone cable

Available Channels:
72MHz: 17 wide band, 40 narrow band
216MHz: 19 wide band, 38 narrow band
(Modulation is automatically switched when changing between narrow and wide band channels)

Physical Dimensions:
3"W x 1"D x 5"H
(7.6cm W x 2.5cm D x 13cm H)

Weight:
< 1 lb. (0.45 kg) with batteries

Door:
Channel and ON/OFF buttons are accessible through the door, lockable

Power:
Two AA batteries, alkaline or NiMH rechargeable; external power connector 8-9 VDC, center positive, <300mA

Battery Life:
16 hours with high capacity alkaline
8 hours with rechargeable NiMH

Battery Charging:
Fully automatic, < 14 hours

Phantom Power:
Included for microphone

Microphone Input:
-55dBm, 50 ohms, unbalanced, tip of 3.5MM connector

Line Input:
-10dBm, 10k ohms, unbalanced, ring of 3.5MM connector

Audio Processing:
40dB automatic 3-speed dynamic control with noise gate

Controls:
On/Off/Program/Lockout, Channel UP, Channel DOWN, Battery Type

Mute Control:
Located on top of unit, LED flashes when mute is activated

Programmability:
Allows only programmed channels to be tuned and to be displayed

LCD:
Channel, Battery/Charge Status, Program, Locked, Channel Lock Out

LED:
Red, Constant = ON and/or charged, fast flash = muted, slow flash = battery low or charging

Specifications subject to change without notice.

Listen Receivers:
LR-500 Programmable Display Receiver,
LR-400 Display Receiver and
LR-300 Digital Receiver

Differences in the LR- 300, 400 and 500

The LR-300 does not have a LCD Display, is only avaiable in the 72MHz band, accesses 17 wide band channels and uses a SEEK button to change channels. Audio quality and charging capabilities are the same.

Differences between the LR-500 and LR-400 include: SEEK and channel UP and DOWN buttons are behind the door on the LR-400 versus protruding through the door on the LR-500. Also, the LR-500 is programmable. See page 25 for LR-500 programming features. (Other than these differences, instructions for use of Listen's LR-400 and LR-500 receivers and their specifications are identical.)

The LR-400 has channel UP, DOWN and SEEK buttons behind the door.

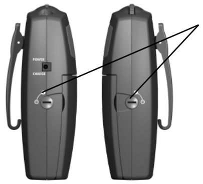
The LR-300 has a SEEK button only and it is located behind the front door.

The LR-500 has channel UP, DOWN and SEEK buttons protruding through the door.

Preparing Listen Receivers for Use

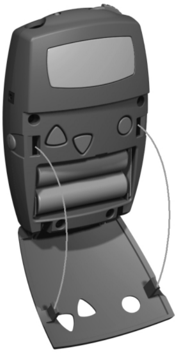
1 **Remove the product from the shipping container** and plastic cover. Inspect for physical damage. See "Technical Service Contacts" (Page 4) if damage has occurred.

2 **Check to see if the door is locked.** If locked, use a pocketknife or small screwdriver to unlock the door locks on the both sides of the unit. To unlock the door, rotate the lock 1/4 turn counter clockwise.



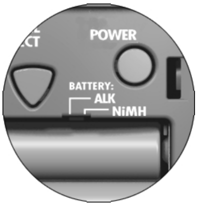
To unlock door: On both the right and left side, unlock by turning slot horizontal as shown.

3 **Open the front access** door by gripping the two tabs with your thumb and index finger.



4 **Select Battery Type** – **CAUTION:** If you are using any battery type other than rechargeable Nickel Metal Hydride (NiMH) batteries, make sure the BATTERY SELECT switch is in the **alkaline** position (located in the battery compartment and labeled BATTERY SELECT, remove batteries to access this switch). The product is shipped from the factory in the alkaline position.

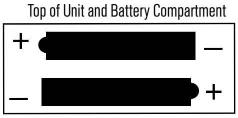
To use NiMH batteries, before placing in unit, change the battery switch to NiMH.



BATTERY SELECT Switch - With batteries removed, use a pen or small screwdriver to select the battery type

WARNING: Do not place the BATTERY SELECT switch in the NiMH position when using Non-Nickel Metal Hydride batteries. The NiMH position will attempt to charge the batteries. Charging non-Nickel Metal Hydride batteries may result in physical harm, destruction of property and/or fire.

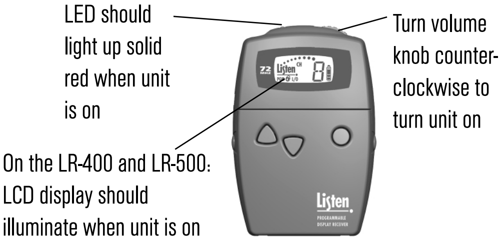
5 **Place Batteries in Unit** Place two AA batteries in the compartment, making note of the battery polarity shown in the battery compartment, and again verifying that the BATTERY SELECT switch is switched correctly for your batteries (see Step 4).


Top of Unit and Battery Compartment

6 **Place an earpiece or headset in the mini jack** on the top of the unit. The LR-500 and LR-400 are designed to work with either mono or stereo connectors.

NOTE: Listen receivers accept mono or stereo jacks.

7 **Turn the receiver on** by rotating the volume knob counter-clockwise. The red LED should activate and, on the LR-400 and LR-500, the LCD display should illuminate. If they do not, check to ensure you have installed charged batteries and that the batteries have been installed with the proper polarity.


LED should light up solid red when unit is on
Turn volume knob counter-clockwise to turn unit on
On the LR-400 and LR-500: LCD display should illuminate when unit is on

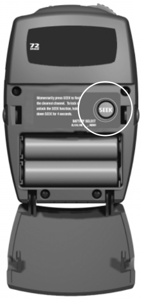
Operating Listen Receivers

1 **Make sure the unit is on** by rotating the volume knob to ON when an earpiece or headset is plugged into the unit. Listen receivers use the cable of the earphone or headset as a receiving antenna.

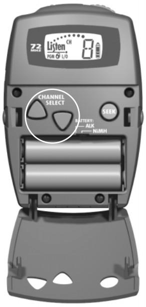
CAUTION: Don't blow out your ears! Slightly rotate the volume knob to turn the unit ON while keeping the volume turned down. Then, adjust volume after listening to earphone.

2 **Select channel:** match the receiver's channel to the transmission channel. For the LR-300, follow instructions on 2a. For the LR-400 and LR-500, follow instructions on 2b.

2a) LR-300 channel selection: first, make sure the transmitter is transmitting continuous audio within range. Second, place the receiver's earphone on your ear to listen. Turn on and adjust volume to reasonable level. Open the LR-300 receiver's front access door and press the SEEK button for less than one second. Listen. Do this three or more times and decide which audio channel sounds best because you won't have channel verification otherwise. Continue to do this until you get back to the best sounding audio channel. You can electronically lock it into the selected channel by pressing and holding the same SEEK button for five or more seconds. (Press and hold the SEEK button again for five+ seconds to unlock.) You're set. Close the front access door.



2b) LR-400 and LR-500 Display Receivers: Select the channel to match the transmission channel by pressing the UP and DOWN buttons.



At 72MHz, each LR-400 and LR-500 receiver operates on 17 wide band channels and 40 narrow band channels. (Letters = Wide Band Channels, Numbers = Narrow Band Channels)

At 216MHz: receivers operate on 19 wide band channels and 38 narrow bands: Channels starting with "2" are wide band, and channels beginning with "1" or "3" are narrow band channels.

Please refer to the Frequency Compatibility Tables section (Pages 28-31) for specific frequencies and compatibility with other manufacturers.

■ **NOTE: Operating on a narrow band channel will have more noise.**

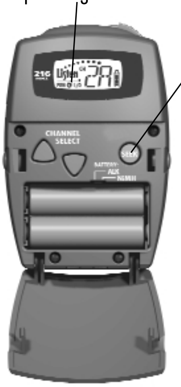
2b continued (LR-400 and LR-500): Test the audio. If a transmitter is broadcasting on that channel, you will be able to hear the audio. If the signal is too weak, the audio will be muted (squelched).

2b continued (LR-400 and LR-500): Find audio transmission. Another way to select a channel on the LR-400 and LR-500 is to press the SEEK button. When you do this, the Listen receiver looks for the next channel that is active. Sometimes the unit will mistake interference for a real broadcast signal. If you get interference, simply press the SEEK button again. The SEEK button will also likely stop on adjacent narrow band channels, or the wide band channel associated with the narrow band you are looking for. Simply continue to press SEEK until you reach the clearest operating channel.

3 **Adjust volume** control for the best listening level.

Locking into Only One Channel

On the LR-400 and LR-500, when locked, the unit will display the LOCK symbol. On the LR-300 (no LCD display), when listening to the unit, it will sound like the channel isn't changing when pressing the SEEK button, thus it is electronically locked.



To electronically lock in the channel so it cannot be changed, press the SEEK button for five seconds or, on the LR-400 and LR-500, until the LOCK symbol is displayed.

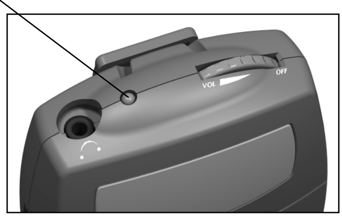
To electronically unlock the channel, press and hold down the SEEK button for five seconds or, on the LR-400 and LR-500, until the LOCK symbol disappears on the display.

Receiver LED Indicator

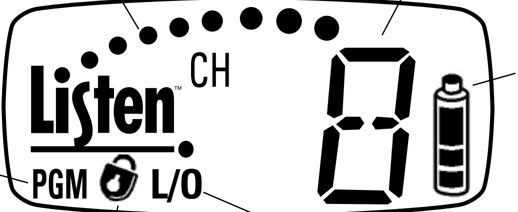
Steady Red LED = Proper Operation

Slow Flashing = Battery is Low

Slow Flashing when Charging = Unit is Charging



LR-400 and LR-500 Receivers: Look & Listen™ Display Descriptions



Small dots indicate the relative RF signal strength. When there is no signal the unit is squelched

Letters and numbers indicate the channel the unit is tuned to

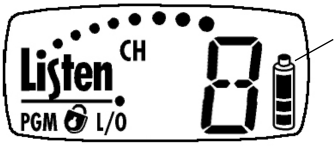
Battery indicator – flashes slowing when charging, quickly when almost discharged

Indicates the unit is in PROGRAM mode (to enter PROGRAM mode, press UP and DOWN until the PGM symbol on the LCD display is shown)

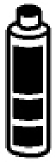
In OPERATING mode, this icon indicates the channel is locked and cannot be changed by the user (to lock/unlock, press the SEEK button until the LOCK symbol on the LCD display is displayed / not displayed)

In PROGRAM mode (PGM symbol is displayed), icon indicates the channel is locked out and the user does not have access to the channel

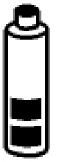
LR-400 and LR-500 Receiver Battery Status Icon



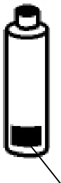
The status of the battery is indicated by the battery symbol on the LCD.



When the battery is fully charged, all three segments of the battery symbol will be displayed, as shown.



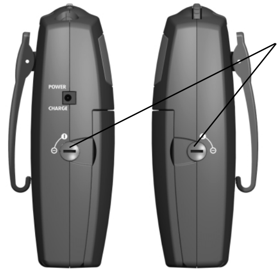
As fewer battery segments are displayed, the lower the battery level, as shown.



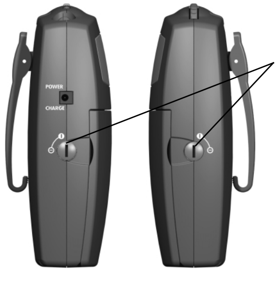
When the batteries are almost fully discharged, the battery segments and the red LED near the volume control will blink. When this occurs, less than 5% of the battery life is remaining and the batteries should be immediately replaced or recharged.

This segment will blink when batteries are low.

Locking the Receiver's Access Door



To unlock: On both the right and left side, unlock by turning slot horizontal as shown.



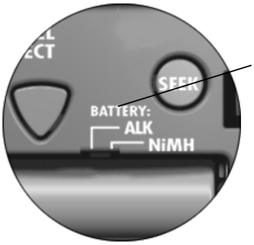
To lock: On both the right and left side, lock door by turning slot vertical as shown.

All Receivers: Charging Batteries Using SmartCharge™

1. Listen's receivers are unique because they have an automatic battery charger built in. That means that when the Listen receiver is connected to a wall transformer or placed in a Listen battery charging case (LA-311 or LA-317), Nickel Metal Hydride (NiMH) batteries will be charged while in the receiver. Once batteries are charged, the internal battery charger will automatically turn off. This prevents overcharging of batteries, automates the process of charging, and makes it so you don't have to remember to unplug the charger.
2. Do not attempt to charge alkaline or any other type of non-NiMH batteries. Only charge NiMH batteries within Listen receivers. We recommend that you use NiMH batteries supplied by Listen.

WARNING: Attempting to charge non-NiMH batteries may result in physical harm, destruction of property and /or fire.

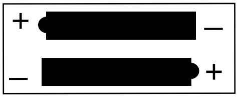
3. **IMPORTANT:** To charge NiMH batteries, the BATTERY SELECT switch must be selected to NiMH. Use a pen or small screwdriver to the move the switch (located in the battery compartment directly below the BATTERY SELECT label) to the proper position.



For only NiMH batteries: set BATTERY SELECT switch to NiMH. For all other batteries, set BATTERY SELECT at alkaline.

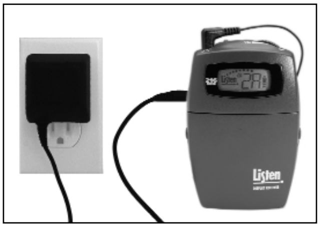
4. Install NiMH rechargeable batteries in the battery compartment ensuring that proper polarity has been observed.

Top of Unit and Battery Compartment



Charging the Batteries (continued)

5. Plug a wall transformer into the side of the Listen receiver or place receiver in a Listen Carrying/Charging Case (LA-311 or LA-317).



For charging using a transformer: with NiMH batteries in unit, plug wall transformer in CHARGING connection and into wall outlet (as shown).

Batteries will be fully charged when red LED stops flashing and is unlit. (LED and battery symbol will flash slowly while charging.)

6. Once the batteries are fully charged, the unit will stop charging the batteries. Full-charge will be indicated when all the battery symbol segments are illuminated and the LED near the volume control is not lit.

7. The unit can be operated while batteries are charging.

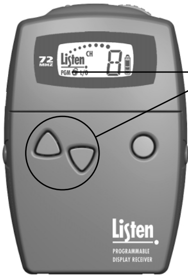
Operating the Receiver with a Wall Transformer

1. The unit will operate normally when connected to a wall transformer.
2. Batteries do not need to be installed in the unit operate from a wall transformer.

LR-500 Receiver Programming

1. The LR-500 can be programmed to operate on a limited number of channels. (The LR-400 and LR-300 cannot be programmed.) For applications where users are required to select a channel (such as language interpretation or classrooms), and you don't want them to have to go through all of the available channels, this feature is ideal. If all channels where available, the user could have 57 channels to select from.

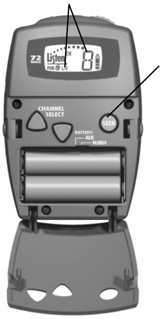
2.



To enter PROGRAM mode, press the UP and DOWN keys simultaneously until the PGM symbol is displayed on the LCD display.

3. **Use the UP and DOWN keys to select channels.** As channels are displayed, the L/O symbol on the LCD display will be displayed for those channels that are locked out. When a channel is locked out, the channel is not available to the user. To toggle between locked out and not locked out, press the SEEK button.

On each channel, check to see if "L/O" is displayed; it means channel is locked out from the user



To toggle between locked out and not locked out, press the SEEK button.

4. **To adjust squelch,** the padlock icon will appear after you program your channels in PROGRAM mode. You should still be in PROGRAM mode. To NOT adjust squelch, don't press a button for 4 seconds and the unit will exit PROGRAM mode. To adjust squelch, press SEEK to unlock squelch (the padlock will disappear). The squelch level will appear as a number of dots showing. No dots means no squelch. The factory default setting is four dots. Press the UP button to increment through five possible squelch settings indicated by dots. More dots means more squelch sensitivity.

In PROGRAM mode on the LR-500, dots indicate level of squelch

Listen
PGM
L/O
Listen
PROGRAMMABLE DISPLAY RECEIVER

5. **To exit PROGRAM mode,** don't press a key for 4 seconds.

LR-500 Specifications

RF Frequency Range:

72MHz: 72.05 - 75.95 MHz
216MHz: 216.025 - 216.975 MHz

Sensitivity:

.6uV typical, 1uV maximum for 12dB SINAD, Max deviation at ±70kHz

Signal to Noise ratio:

72MHz: Wide band channels, 60dB;
Narrow, 54dB
216MHz: Wide band channels, 50dB;
Narrow, 44dB

Frequency Response:

72 50Hz to 17kHz, ±3dB
216 50Hz to 10kHz, ±3dB

Distortion:

< 2% THD

Squelch:

40dB with loss of RF signal

Antenna:

Uses headphone cable

Available Channels:

72MHz: 17 wide band, 40 narrow band
216MHz: 19 wide band, 38 narrow band
(see charts on pages 28-31 for details)

Physical Dimensions:

3" W x 1"D x 5"H (7.6cm W x 2.5cm D x 13cm H)

Weight:

< 1 lb. (.45 kg) with batteries

Door:

CHANNEL and SEEK buttons are accessible through door, lockable

LR-500 Programmability:

Allows only programmed channels to appear on the display, user functions can be locked by pressing the Seek button for > 3 seconds

Power:

Two AA batteries, high capacity alkaline or NiMH rechargeable; external power connector 8-9 VDC, tip positive,<300mA

Battery Charging:

Fully automatic, <14 hours

Battery Life:

40 hours with high capacity alkaline,
20 hours with rechargeable NiMH

Headphone Output:

750mW RMS maximum at 25kHz deviation, 32 ohms, 3.5MM mono or stereo

LR-400 Specifications

RF Frequency Range:

72MHz: 72.05 - 75.95 MHz
216MHz: 216.025 - 216.975 MHz

Sensitivity:

.6uV typical, 1uV maximum for 12dB SINAD, Max deviation at ±70kHz

Signal to Noise ratio:

72MHz: Wide band channels, 60dB;
Narrow, 54dB
216MHz: Wide band channels, 50dB;
Narrow, 44dB

Frequency Response:

72 50Hz to 17kHz, ±3dB
216 50Hz to 10kHz, ±3dB

Distortion:

< 2% THD

Squelch:

40dB with loss of RF signal

Antenna:

Uses headphone cable

Available Channels:

72MHz: 17 wide band, 40 narrow band
216MHz: 19 wide band, 38 narrow band
(see charts on pages 28-31 for details)

Physical Dimensions:

3" W x 1"D x 5"H (7.6cm W x 2.5cm D x 13cm H)

Weight:

< 1 lb. (.45 kg) with batteries

Door:

CHANNEL and SEEK buttons are accessible through door, lockable

Power:

Two AA batteries, high capacity alkaline or NiMH rechargeable; external power connector 8-9 VDC, tip positive, <300mA

Battery Charging:

Fully automatic, <14 hours

Battery Life:

40 hours with high capacity alkaline,
20 hours with rechargeable NiMH

Headphone Output:

750mW RMS maximum at 25kHz deviation, 32 ohms, 3.5MM mono or stereo

LR-300 Specifications

RF Frequency Range:

72MHz: 72.05 - 75.95 MHz

Sensitivity:

.6uV typical, 1uV maximum for 12dB SINAD, Max deviation at ±70kHz

Signal to Noise ratio:

72MHz: Wide band channels, 60dB

Frequency Response:

50Hz to 17kHz, ±3dB

Distortion:

< 2% THD

Squelch:

40dB with loss of RF signal

Antenna:

Uses headphone cable

Available Channels:

72MHz: 17 wide band

Frequency	Channel Letter
72.100	A
72.200	K
72.300	B
72.400	N
72.500	C
72.600	O
72.700	D
72.800	P
72.900	E
74.700	I
75.300	J
75.400	R
75.500	F
75.600	S
75.700	G
75.800	T
75.900	H

Physical Dimensions:

3" W x 1"D x 5"H (7.6cm W x 2.5cm D x 13cm H)

Weight:

< 1 lb. (.45 kg) with batteries

Door:

CHANNEL and SEEK buttons are accessible through door, lockable

Power:

Two AA batteries, high capacity alkaline or NiMH rechargeable; external power connector 8-9 VDC, tip positive,<300mA

Battery Charging:

Fully automatic, <14 hours

Battery Life:

40 hours with high capacity alkaline,
20 hours with rechargeable NiMH

Headphone Output:

750mW RMS maximum at 25kHz deviation, 32 ohms, 3.5MM mono or stereo

Specifications subject to change without notice.

LR-300, LR-400 and LR-500 Receiver Troubleshooting

The receiver has no power.

Make sure the unit has either fully charged batteries or a Listen approved wall transformer connected. Make sure the VOLUME knob has been rotated to the ON position. If this does not work, try a different set of batteries. Make sure the batteries are installed properly.

There is no audio.

Make sure you have turned the volume control up. Make sure the earphone jack is plugged all the way in to the top of unit. Make sure the transmitter is broadcasting an audio source. Make sure you're tuned to the same channel as the transmitter. If the RF signal is too weak, the receiver will squelch and thus mute the audio source; move closer to the antenna or make sure the transmitter's output RF power select is on 'FULL' .

The audio is distorted.

Make sure you're receiving on the correct channel. Make sure the audio on the transmitter is not turned up too loud so that it is clipping or distorting. Make sure the earphone connector is pushed all the way in. If you're using the LR-300 (without an LCD display), re-scan the channels to find the clearest channel by pressing SEEK (see page 22, section 2a for more information).

There is interference.

Try different frequencies until you find a clear channel. If this does not work, try a different frequency band (i.e. if you're using 72MHz, try 216MHz or vice versa). This is accomplished by returning the equipment to Listen (at no charge).

I cannot pick up the signal on the receiver.

Check to make sure the receiver and the transmitter are on the same frequency. Also make sure the receiver is in broadcast range of the transmitter.

I can pick up the signal on the receiver, but it sounds like it's not tuned in.

Check to make sure the transmitter and receiver are on exactly the same channel number/letter. If you're using the LR-300 (without an LCD display), re-scan the channels to find the clearest channel by pressing SEEK (see page 22, section 2a for more information).

I'm using another brand of transmitter - how do I tell which channel to use?

Refer to Listen's Frequency Compatibility Tables (included on pages 28-31). Either adjust the transmitter or the receiver to a common channel.

There is not sufficient range.

Make sure you are located as close to the antenna of the transmitter as possible. Try to locate the antenna as high as possible and free from obstacles. You can also check the squelch setting; perhaps it's too sensitive. To change this setting, refer to instructions on page 25, #4 under programming.

It's confusing for users to have 57 channels when switching between channels.

Use the programming feature of the LR-500 (only) to limit the number of channels users have access to.

Users keep changing channels.

You can prevent users from changing channels by pressing the SEEK button and holding it for five seconds. This will electronically lock into the channel. In addition, lock the access door to prevent users from accessing channel buttons.

I cannot change channels when pressing the UP and DOWN buttons or the SEEK button.

The unit is electronically locked. Press the SEEK button for five seconds to unlock.

When I change channels, only certain channels are accessible.

The unit has been programmed to tune to only certain channels (LR–500 ONLY!). You can change these tunable channels by re-programming the unit.

How can I limit access to the batteries?

Use the side door locks.

My batteries are not charging.

Make sure the battery switch is in the NiMH position. Make sure you have connected an approved Listen charging transformer. Make sure you use ONLY NiMH batteries provided by Listen. Never try to charge alkaline or NICAD batteries.

I want to run the unit from a wall transformer.

Simply plug a Listen approved transformer to the CHARGE/POWER connector. Batteries need not be installed, however, NiMH will recharge if installed while the unit is being used.

Listen™

72MHz Frequency Compatibility Table

Frequency (MHz)	LTC	Phonic Ear	Comtek	Williams	Telex	Drake	GTNR
72.025	1	1	1	11			
72.050							1
72.075	2	2	2	12			
72.100	A	A	A	A, 13	A	72.1	2
72.125	3	3	3	14			
72.150							3
72.175	4	4	4	15			
72.200	K				B	72.2	4
72.225	5	5	5	16			
72.250							5
72.275	6	6	6	17			
72.300	B	B	B	B, 18	C	72.3	6
72.325	7	7	7	19			
72.350							7
72.375	8	8	8	20			
72.400	N				D	72.4	8
72.425	9	9	9	21			
72.450							9
72.475	10	10	10	22			
72.500	C	C	C	C, 23	E	72.5	10
72.525	11	11	11	24			
72.550							11
72.575	12	12	12	25			
72.600	O				F	72.6	12
72.625	13	13	13	26			
72.650							13
72.675	14	14	14	27			
72.700	D	D	D	D, 28	G	72.7	14
72.725	15	15	15	29			
72.750							15
72.775	16	16	16	30			
72.800	P				H	72.8	16
72.825	17	17	17	31			
72.850							17
72.875	18	18	18	32			
72.900	E	E	E	E, 33	I	72.9	18
72.925	19	19	19	34			
72.950							19
72.975	20	20	20	35			
74.625	33	33	33	36			
74.650							20

72MHz Frequency Compatibility Table (continued)

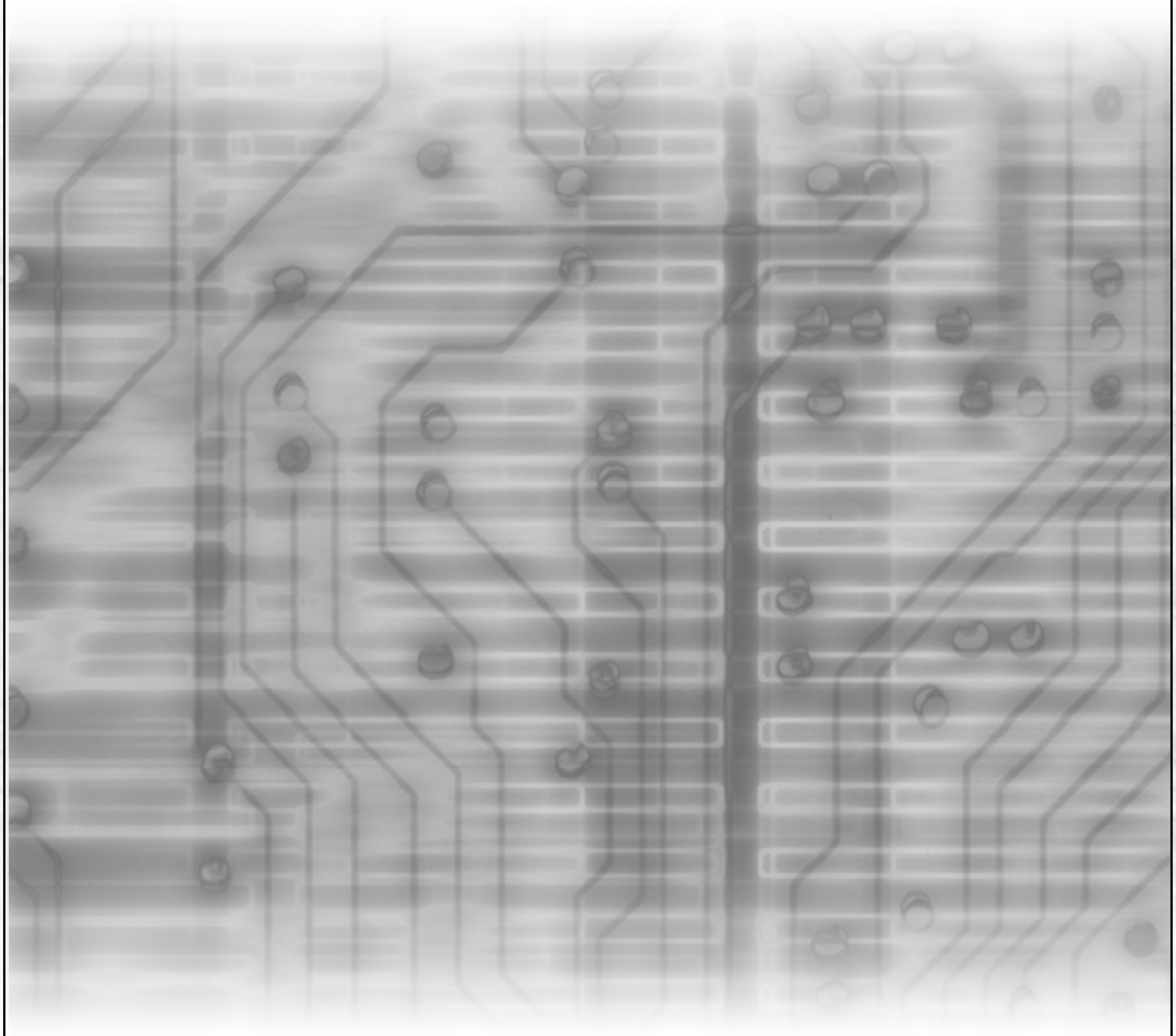
Frequency (MHz)	LTC	Phonic Ear	Comtek	Williams	Telex	Drake	GTNR
74.675	34	34	34	37			
74.700	I	I	I	I, 38	O		21
74.725	35	35	35	39			
74.750							22
74.775	36	36	36	40			
75.225	37	37	37	41			
75.250							23
75.275	38	38	38	42			
75.300	J	J	J	J, 43	P		24
75.325	39	39	39	44			
75.350							25
75.375	40	40	40	45			
75.400	R				Q		26
75.425	21	21	21	46			
75.450							27
75.475	22	22	22	47			
75.500	F	F	F	F, 48	J	75.5	28
75.525	23	23	23	49			
75.550							29
75.575	24	24	24	50			
75.600	S				K	75.6	30
75.625	25	25	25	51			
75.650							31
75.675	26	26	26	52			
75.700	G	G	G	G, 53	L	75.7	32
75.725	27	27	27	54			
75.750							33
75.775	28	28	28	55			
75.800	T				M	75.8	34
75.825	29	29	29	56			
75.850							35
75.875	30	30	30	57			
75.900	H	H	H	H, 58	N	75.9	36
75.925	31	31	31	59			
75.950							37
75.975	32	32	32	60			

216MHz Frequency Compatibility Table

Frequency (MHz)	LTC	Williams	CSI	Comtek	Phonic E	Telex	Drake	Gentner
216.5250	2K	H	3					10
216.5375	3K							
216.5500	x							
216.5625	1L							
216.5750	2L	I	12					11
216.5875	3L							
216.6000	x							
216.6125	1M							
216.6250	2M	J	8					12
216.6375	3M							
216.6500	x							
216.6625	1N							
216.6750	2N	K	16					13
216.6875	3N							
216.7000	x							
216.7125	1P							
216.7250	2P	L	19					14
216.7375	3P							
216.7500	x							
216.7625	1R							
216.7750	2R		4					15
216.7875	3R							
216.8000	x							
216.8125	1S							
216.8250	2S		13					16
216.8375	3S							
216.8500	x							
216.8625	1T							
216.8750	2T		9					17
216.8875	3T							
216.9000	x							
216.9125	1U							
216.9250	2U		17					18
216.9375	3U							
216.9500	x							
216.9625	1V							
216.9750	2V		5					19
216.9875	3V							
217.0000	x							

Frequency (MHz)	LTC	Williams	CSI	Comtek	Phonic E	Telex	Drake	Gentner
216.5250	2K	H	3					10
216.5375	3K							
216.5500	x							
216.5625	1L							
216.5750	2L	I	12					11
216.5875	3L							
216.6000	x							
216.6125	1M							
216.6250	2M	J	8					12
216.6375	3M							
216.6500	x							
216.6625	1N							
216.6750	2N	K	16					13
216.6875	3N							
216.7000	x							
216.7125	1P							
216.7250	2P	L	19					14
216.7375	3P							
216.7500	x							
216.7625	1R							
216.7750	2R		4					15
216.7875	3R							
216.8000	x							
216.8125	1S							
216.8250	2S		13					16
216.8375	3S							
216.8500	x							
216.8625	1T							
216.8750	2T		9					17
216.8875	3T							
216.9000	x							
216.9125	1U							
216.9250	2U		17					18
216.9375	3U							
216.9500	x							
216.9625	1V							
216.9750	2V		5					19
216.9875	3V							
217.0000	x							

User Manual Digitool™ MX



ARCHITECTURAL ACOUSTICS™

By Peavey

Listen to This™

Intended to alert the user to the presence of uninsulated “dangerous voltage” within the product’s enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons.

Intended to alert the user of the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

CAUTION: Risk of electrical shock — DO NOT OPEN!
CAUTION: To reduce the risk of electric shock, do not remove cover. No user serviceable parts inside. Refer servicing to qualified service personnel.

WARNING: To prevent electrical shock or fire hazard, do not expose this appliance to rain or moisture. Before using this appliance, read the operating guide for further warnings.

Este símbolo tiene el propósito, de alertar al usuario de la presencia de “(voltaje) peligroso” sin aislamiento dentro de la caja del producto y que puede tener una magnitud suficiente como para constituir riesgo de descarga eléctrica.

Este símbolo tiene el propósito de alertar al usario de la presencia de instruccones importantes sobre la operación y mantenimiento en la información que viene con el producto.

PRECAUCION: Riesgo de descarga eléctrica ¡NO ABRIR!
PRECAUCION: Para disminuír el riesgo de descarga eléctrica, no abra la cubierta. No hay piezas útiles dentro. Deje todo mantenimiento en manos del personal técnico cualificado.

ADVERTENCIA: Para evitar descargas eléctricas o peligro de incendio, no deje expuesto a la lluvia o humedad este aparato Antes de usar este aparato, lea más advertencias en la guía de operación.

Ce symbole est utilisé dans ce manuel pour indiquer à l’utilisateur la présence d’une tension dangereuse pouvant être d’amplitude suffisante pour constituer un risque de choc électrique.

Ce symbole est utilisé dans ce manuel pour indiquer à l’utilisateur qu’il ou qu’elle trouvera d’importantes instructions concernant l’utilisation et l’entretien de l’appareil dans le paragraphe signalé.

ATTENTION: Risques de choc électrique — NE PAS OUVRIR!
ATTENTION: Afin de réduire le risque de choc électrique, ne pas enlever le couvercle. Il ne se trouve à l’intérieur aucune pièce pouvant être reparée par l’utilisateur. Confiez l’entretien et la réparation de l’appareil à un réparateur Peavey agréé.

AVERTISSEMENT: Afin de prévenir les risques de décharge électrique ou de feu, n’exposez pas cet appareil à la pluie ou à l’humidité. Avant d’utiliser cet appareil, lisez attentivement les avertissements supplémentaires de ce manuel.

Dieses Symbol soll den Anwender vor unisolierten gefährlichen Spannungen innerhalb des Gehäuses warnen, die von Ausreichender Stärke sind, um einen elektrischen Schlag verursachen zu können.

Dieses Symbol soll den Benutzer auf wichtige Instruktionen in der Bedienungsanleitung aufmerksam machen, die Handhabung und Wartung des Produkts betreffen.

VORSICHT: Risiko — Elektrischer Schlag! Nicht öffnen!
VORSICHT: Um das Risiko eines elektrischen Schlages zu vermeiden, nicht die Abdeckung enfernen. Es befinden sich keine Teile darin, die vom Anwender repariert werden könnten. Reparaturen nur von qualifiziertem Fachpersonal durchführen lassen.

ACHTUNG: Um einen elektrischen Schlag oder Feuergefahr zu vermeiden, sollte dieses Gerät nicht dem Regen oder Feuchtigkeit ausgesetzt werden. Vor Inbetriebnahme unbedingt die Bedienungsanleitung lesen.

IMPORTANT SAFETY INSTRUCTIONS

WARNING: When using electric products, basic cautions should always be followed, including the following:

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water. For example, near or in a bathtub, swimming pool, sink, wet basement, etc.
6. Clean only with a dry cloth.
7. Do not block any of the ventilation openings. Install in accordance with manufacturer's instructions. It should not be placed flat against a wall or placed in a built-in enclosure that will impede the flow of cooling air.
8. Do not install near any heat sources such as radiators, heat registers, stoves or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding plug. The wide blade or third prong is provided for your safety. When the provided plug does not fit into your inlet, consult an electrician for replacement of the obsolete outlet. Never break off the grounding. Write for our free booklet "Shock Hazard and Grounding." Connect only to a power supply of the type marked on the unit adjacent to the power supply cord.
10. Protect the power cord from being walked on or pinched, particularly at plugs, convenience receptacles, and the point they exit from the apparatus.
11. Only use attachments/accessories provided by the manufacturer.
12. Use only with a cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
13. Unplug this apparatus during lightning storms or when unused for long periods of time.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.
15. If this product is to be mounted in an equipment rack, rear support should be provided.
16. Exposure to extremely high noise levels may cause a permanent hearing loss. Individuals vary considerably in susceptibility to noise-induced hearing loss, but nearly everyone will lose some hearing if exposed to sufficiently intense noise for a sufficient time. The U.S. Government's Occupational and Health Administration (OSHA) has specified the following permissible noise level exposures:

Sound Duration Per Day In Hours	Sound Level dBA, Slow Response
8	90
6	92
4	95
3	97
2	100
1 1/2	102
1	105
1/2	110
1/4 or less	115

According to OSHA, any exposure in excess of the above permissible limits could result in some hearing loss. Ear plugs or protectors to the ear canals or over the ears must be worn when operating this amplification system in order to prevent a permanent hearing loss, if exposure is in excess of the limits as set forth above. To ensure against potentially dangerous exposure to high sound pressure levels, it is recommended that all persons exposed to equipment capable of producing high sound pressure levels such as this amplification system be protected by hearing protectors while this unit is in operation.

SAVE THESE INSTRUCTIONS!

Description

The Digitool™ MX is a fully programmable audio processing and control system. It includes advanced parallel DSP processing, a multi-layer front panel display, and multiple interface and control options. Signal processing is accomplished using two 32-bit parallel SHARC processors with a combined power of 132 Mips. The Digitool MX gives the system designer an economical alternative to design, and is considerably more powerful than similarly priced components.

The Digitool MX's host processor and non-volatile flash memory maintain user-defined configurations and allow interface via the unit's front panel control and optional external control interfaces. Completely upgradable through a standard serial connection to a PC, the Digitool MX allows the system designer to upgrade to the latest version without removing the unit from the rack and disassembling the chassis.

The graphic display is capable of displaying real-time audio signal activity, active presets, and control information for audio processes, as well as monitoring critical variables such as internal cabinet temperature and power presence.

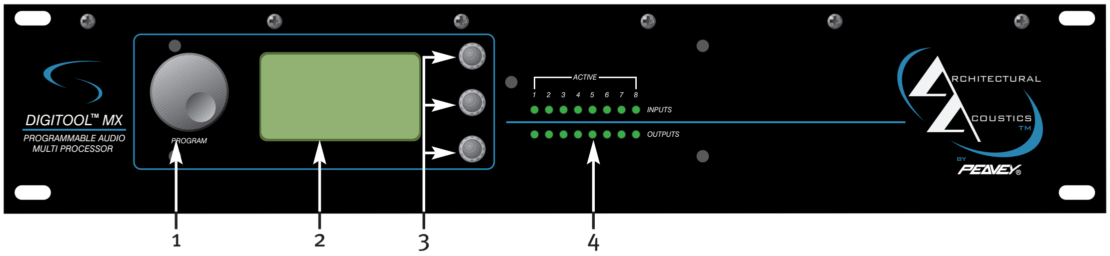
Features

- Eight balanced analog line/mic inputs
- Phantom power
- Eight balanced analog line outputs
- 24-bit A/D and D/A converters
- Up to 96 kHz sample rate
- Sixteen front panel dual-color LED indicators (eight input, eight output)
- Three front panel parameter controls
- Large 64 x 128 pixel display with back light
- Jog/select control encoder for navigating the display
- 32-bit parallel processing performed by two ADI SHARC DSPs
- RS-232 connection for easy set-up and operation
- RS-485 connection for remote serial functions
- Four assignable control input ports(analog) with 0-10 VDC range for external potentiometers
- Worldwide-compatible power supply

Applications

- Stadiums
- Auditoriums
- Civic Centers
- Performing Arts Centers
- Theaters
- Courts of Law
- Houses of Worship
- Theme Parks
- Arenas
- University Campus Buildings
- Music Clubs
- Hotel Meeting Rooms
- Conference Centers
- Schools
- Sports Bars
- Cruise Ships
- Teleconferencing
- Paging Systems
- Any facility requiring distribution of multiple line-level signals

Front Panel


DIGITOOL™ MX
PROGRAMMABLE AUDIO MULTI PROCESSOR
PROGRAM
ACTIVE
1 2 3 4 5 6 7 8
INPUTS
OUTPUTS
ARCHITECTURAL ACOUSTICS TM
BY PEAVEY
1
2
3
4

1. **PROGRAM WHEEL:**

   This control navigates the display as well as performing parameter changes on functions with more than three settable parameters. It is used to scroll through the various functions, and to select among items such as inputs/outputs and hardware configuration.

2. **LCD DISPLAY:**

   This screen gives a graphical representation of unit configuration and parameter settings. The layout of the screen and the information conveyed will depend on the function being monitored or set.

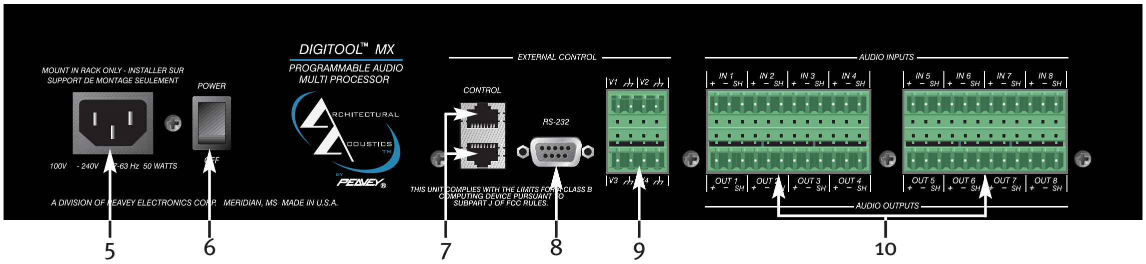
3. **PARAMETER CONTROLS**:

   These controls are used to adjust the settings of the displayed effect. Their individual function will depend on the effect being displayed.

4. **ACTIVE LEDs**:

   These LEDs (eight input and eight output) will illuminate as an indication of the active inputs and outputs.

Rear Panel


MOUNT IN RACK ONLY - INSTALLER SUR SUPPORT DE MONTAGE SEULEMENT
POWER
OFF
100V - 240V 47-63 Hz 50 WATTS
A DIVISION OF PEAVEY ELECTRONICS CORP. MERIDIAN, MS MADE IN U.S.A.
DIGITOOL™ MX
PROGRAMMABLE AUDIO MULTI PROCESSOR
ARCHITECTURAL ACOUSTICS TM
BY PEAVEY
EXTERNAL CONTROL
CONTROL
RS-232
THIS UNIT COMPLIES WITH THE LIMITS FOR A CLASS B COMPUTING DEVICE PURSUANT TO SUBPART J OF FCC RULES.
V1 V2 V3 V4
AUDIO INPUTS
IN 1 IN 2 IN 3 IN 4 IN 5 IN 6 IN 7 IN 8
+ − SH
OUT 1 OUT 2 OUT 3 OUT 4 OUT 5 OUT 6 OUT 7 OUT 8
AUDIO OUTPUTS
5
6
7
8
9
10

5. **IEC CONNECTOR**:

   This receptacle is for the IEC line cord (included) that provides AC power to the unit.

   Never break off the ground pin on any equipment. It is provided for your safety.

   If the outlet used does not have a ground pin, a suitable grounding adapter should be used and the third wire should be grounded properly. To prevent the risk of shock or fire hazard, always be sure that the mixer and all associated equipment is properly grounded.

Note for UK only

If the colors of the wires in the mains lead of this unit do not correspond with the colored markings identifying the terminals in your plug, proceed as follows: (1) The wire that is colored green and yellow must be connected to the terminal that is marked by the letter E, the earth symbol, colored green, or colored green and yellow. (2) The wire that is colored blue must be connected to the terminal that is marked with the letter N or the color black. (3) The wire that is colored brown must be connected to the terminal that is marked with the letter L or the color red.

6. **POWER SWITCH:**

This rocker switch applies mains power to the unit.

7. **RJ-45 INPUT / OUTPUT CONNECTORS:**

These standard, eight-connector jacks allow connection to a multi-drop RS-485 bus.

8. **RS-232 CONNECTOR:**

This standard DB-9 serial cable connector allows interface between the Digitool™ and a PC.

9. **CONTROL VOLTAGE CONNECTORS:**

These four-control, removable Euro-style input ports (analog) have a 0-10 V DC range to accommodate assignable external potentiometers.

10. **INPUTS AND OUTPUTS:**

These detachable, Euro-style connectors allow analog input to and output from the unit.

NAVIGATING THE DISPLAY

The layout of the front panel is designed to simplify navigation and editing of internal processes. The PROGRAM WHEEL, which is located to the left of the display, controls all cursor movements within the display. The PROGRAM WHEEL has 32 detented positions per full rotation. Clockwise rotation results in left-to-right or top-to-bottom motion while counterclockwise motion reverses this order. Each detent results in the cursor moving by one position. In some view panels, the cursor location remains in a fixed location on the screen while each detent of the wheel results in a view change. This allows the user to navigate through multiple screens in just a few clicks of the PROGRAM WHEEL.

SELECTING DISPLAY OBJECTS

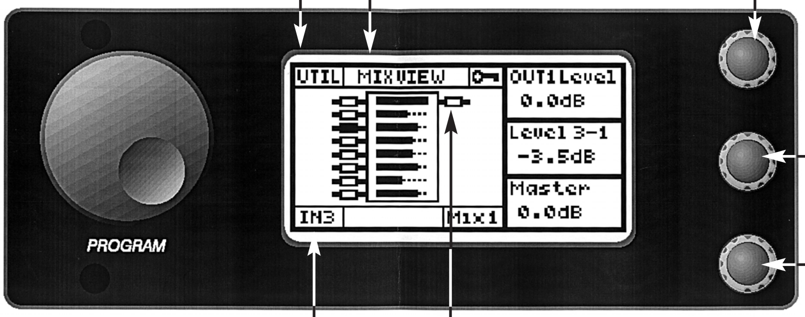
The PROGRAM WHEEL also functions to select items on the screen. If the cursor is over an object on the screen, it may be selected by pressing the PROGRAM WHEEL. This will either take you to an appropriate screen or enable additional options within the display.

PARAMETER CONTROLS

To the right of the display are three undefined encoder controls. The function for these controls are indicated in the display along the right side of the screen, adjacent to the encoders. Figure 1 illustrates the function for each control within the Mix View panel, which is described in the next section.

Panel Screens

The front panel screen consists of four primary views. They are: Mix, Route, Edit and Utilities. The Mix View is the default view. Edit, Route and Utilities are accessed through the Mix view.



Highlighting and selecting the corner box enters the utility view

Mix box provides a quick view of the relative levels of each channel for the selected mix

Controls the level of the selected output

Controls the selected input to output matrix level

Master level control for the entire system

Indicates the currently selected input

Only the selected mix is visible in the mix view. There are eight views, one for each mix

Figure 1. The Mix View is the primary view from which all other functions are accessed.

Mix View

The Mix View is considered to be the main view from which other views and controls are accessed. The Mix View provides a symbolic layout of each of eight output mixes. Figure 1 illustrates the components of this display and identifies the controllable functions. Rotating the PROGRAM WHEEL highlights each input or output channel object, indicated by small rectangles on the left and right sides of the Mix Box. The Mix Box contains bar graphs indicating the levels of all eight inputs that contribute to the selected mix. Along the left side of the Mix Box are the eight inputs. Similar to the input objects in the display are the output objects which represent 1 of 8 mixes. These appear on the right side of the Mix Box. Notice in figure 1 that only one output mix is visible at a time.

MIX VIEW CONTROLS

The three controls for the Mix View are Output Level, Matrix Level and Master Level controls.

OUTPUT LEVEL: The top encoder affects the Output Level for one of eight outputs. This level corresponds to the selected mix along the right side of the Mix Box.

MATRIX LEVEL: The middle encoder controls 1 of 64 levels. This control corresponds to a combination of the selected input channel and output mix. In the illustration of figure 1, channel 3 is highlighted and therefore enables the Matrix Level of channel 3 to be visible for the selected mix. Notice that the nomenclature of the label for this control, which appears in the display adjacent to the encoder, indicates a level control within the 8X8 matrix. That is “Level 3-1” identifies the Input-3 to Mix-1 level. There are 64 active levels within the matrix. All are accessible from the Mix panel view.

MASTER LEVEL: The third or bottom encoder functions as the Master Level for the entire system. When this Level is changed, all eight outputs are changed by an equivalent amount.

Route View

The Route View is an alternative presentation to the Mix View (Figure 2). In the Route View, a single input channel is selected. The display then shows the routing of the selected input channel to each output. Each output level can be changed in the same manner as is done in the Mix View. It should be noted that in either view, routing and mixing can be accomplished. The Route View is handy when you need to route a selected channel to multiple outputs. The three parameter knobs perform the same function as in the Mix View. The top knob controls the highlighted output mix, the second knob controls the selected Matrix Level and the bottom knob controls the Master Volume for all outputs.

Digitool Security

Access to the Digitool's settings may be restricted by clicking on the key icon at the upper right portion of the Mix View screen. An option menu will appear which allows the user to either set the unit's password or to lock the screen.

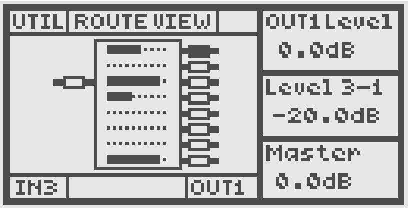
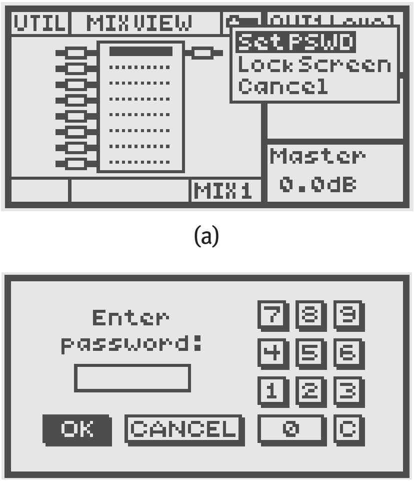

UTIL
ROUTE VIEW
OUT1 Level
0.0dB
Level 3-1
-20.0dB
Master
0.0dB
IN3
OUT1

Figure 2. The Route View allows the user to quickly determine the signals that are mixed to a particular output channel.

In order to set a new password, the user is first asked to enter the old password using the program knob and the displayed graphical numeric keypad. When shipped from the factory, the Digitool's default password is '1234'. After correct entry of the old password, the user will be prompted to enter the new password (up to four digits long), and then to confirm this new password again.

The Digitool will enter secure mode by choosing Lock Screen from the key option menu. In this mode, a banner will appear in the graphics display, and any movements of the front panel controls will cause a password entry screen to appear. A user must type the correct password before they can gain access to the Digitool's settings via the front panel.

If the unit is powered down while in secure mode, then it will power up in secure mode. Similarly, if it was powered down unsecure, it will power up unsecure.


UTIL
MIX VIEW
Set PSWD
Lock Screen
Cancel
Master
0.0dB
MIX1

(a)

Enter password:
7 8 9
4 5 6
1 2 3
OK
CANCEL
0
C

(b)

Figures 3a and 3b. Security options and password entry views.

Edit View - Input Channel

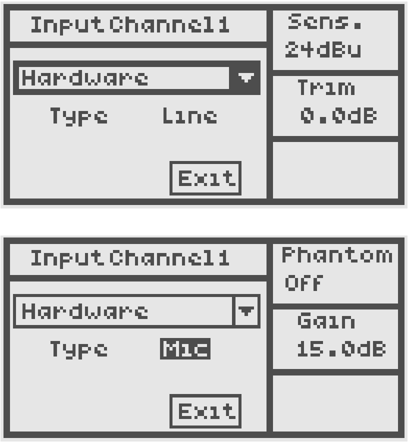
Pressing the **PROGRAM** knob while an input or output object is selected will open the edit window for the selected channel. These panels are used to access and control filters, gates, compressors and other signal processes. Refer to these sections in the following pages of this manual for a detailed description.

Once Edit is selected the Edit View panel will be activated. The Edit View panel is actually a series of views that are presented in the same manner that the signal path follows. The views are described below.

INPUT CHANNEL HARDWARE

The first panel that is visible in Edit View is the Input Channel Hardware page (figures 4a and 4b). This panel allows for configuration of the input hardware. There are two different views depending upon whether Mic or Line is selected.

LINE MODE: The control options for Line mode are Sensitivity and Trim. Sensitivity can be adjusted for one of four settings: 30, 24, 18 and 12 dBu. Each setting scales the full digital range to the specified sensitivity setting. For example, if an input signal has a nominal level of +4 dBu, then a sensitivity setting of 24 dBu provides 20 dB of digital headroom before clip. A compact disc player has a nominal level of –10 dBV (-7.79 dBu) therefore a sensitivity setting of +12 dBu provides 19.8 dB of headroom before clip. Trim provides +/- 6 dB of gain or attenuation in the input hardware.


Input Channel1
Hardware
Type Line
Sens. 24dBu
Trim 0.0dB
Exit

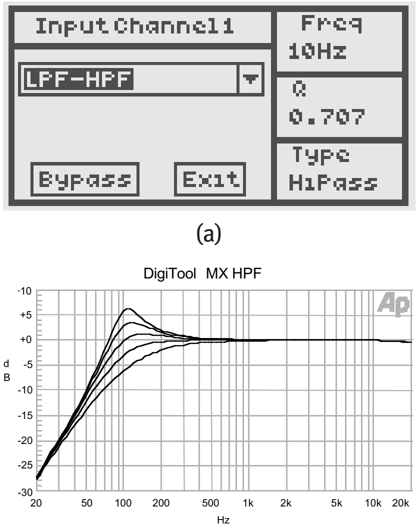
Input Channel1
Hardware
Type Mic
Phantom Off
Gain 15.0dB
Exit

Figures 4a and 4b. Input Channel Hardware views for Line mode (a) and Mic mode (b).

MIC MODE: The control options are Phantom On/Off and Gain. The Mic gain is adjustable from +15 to +50 dB in 0.5 dB steps.

INPUT CHANNEL LPF-HPF

The first DSP process for the input signal chain is the filter. The filter parameters are as follows: Frequency, Q and Type (see figures 5a and 5b). The user can select either a low-pass filter (LPF) or high-pass filter (HPF). In general this filter is intended to be used as a high pass type for the purpose of removing unwanted low frequencies at the input, particularly when used for high-gain microphone applications. This will tend to stabilize dynamic processing such as gates and compressors that appear later in the chain. In addition, the user has control over the Q of the filter.



Input Channel 1

LPF-HPF

Freq 10Hz

Q 0.707

Type HiPass

(a)

(b)

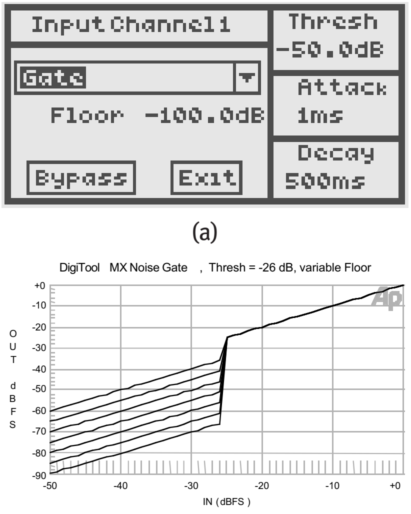
Figures 5a and 5b. Filter control panel (a) and varying Q values (b).

Q ADJUSTMENT: The Q is a primitive component of the filter and is usually fixed in most HPF or LPF implementations. Figure 5b illustrates the effect of the Q adjustment on the frequency response of the filter. The illustration shows the filter in high-pass mode with the corner frequency set for 100 Hz. There are five sweeps shown, each with a different Q value. The normal setting for Q is typically 0.707 which is shown in the illustration as the 2nd sweep from the bottom. At this value, the filter is considered to be critically damped and has a monotonic response throughout the attenuation band. The attenuation at the corner frequency is –3 dB.

Higher Q values result in corner frequency gain rather than attenuation. Lower Q values result in more attenuation at the corner frequency. Note that in either case, the filter still achieves a 12 dB/oct slope below the corner frequency and is ultimately unattenuated above the corner frequency. Notice also that the value of Q specifies the gain at the specified frequency. Therefore a Q of 2 provides a +6 dB boost at the specified corner while a Q of 0.5 attenuates by 6 dB. It is important to observe that the Q value does not specify the maximum boost or cut level of the filter. For example, notice the middle curve which has a Q of 1.0. This curve is 0 dB at the corner frequency yet has slight gain above the corner. One application for this control is to simultaneously roll off the low end and add warmth to the input signal which could serve to free up one of the two parametric equalizers in later stages.

NOISE GATE

The next process in the input signal chain is the Noise Gate. There are four parameter controls for the noise gate. See figures 6a & 6b for the control view and transfer function graph.



Input Channel 1

Gate

Floor -100.0dB

Thresh -50.0dB

Attack 1ms

Decay 500ms

(a)

(b)

Figures 6a and 6b. Panel view (a) and transfer function (b) for the noise gate.

THRESHOLD: The top encoder sets the gate's turn-on threshold. This control is specified in dBFS. The range of the control is from 0 to –100 dBFS.

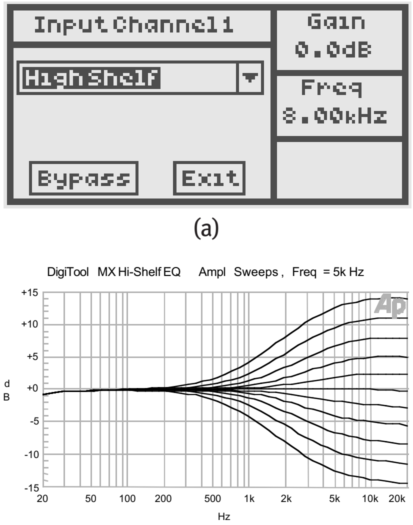
ATTACK: The middle control adjusts the attack time of the gate. The value range is from 1 to 1000 ms in 1 ms steps.

DECAY: The bottom control adjusts the decay time of the gate. The value range is from 1 to 1000 ms in 1 ms steps.

FLOOR: The floor specifies the level to which the gate attenuates when off. This parameter must be accessed and modified using the PROGRAM WHEEL to navigate and select. The range of this control is 0 to –100 dBFS.

HIGH SHELF EQ

This is the first of four shaping EQs for the input signal chain. The top encoder adjusts the Gain (boost/cut) and has a range of –15 dB to +15 dB. The corner frequency is adjustable down to 3 kHz. The corner frequency is positioned 3 dB below the shelf level with a 6 dB/oct slope between the shelf point and 0 dB. The graph of figure 7b shows multiple curves for varying gain settings with the frequency set for 5 kHz.


Input Channel 1

High Shelf

Gain 0.0dB

Freq 8.00kHz

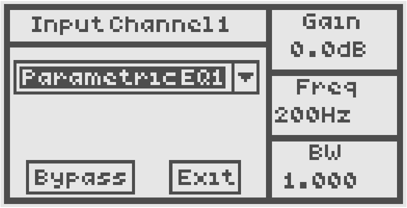
(a)

(b)

Figures 7a and 7b. High Shelving EQ panel view (a) and boost/cut response curves.

PARAMETRIC EQ 1 & 2

The next two processes in the signal chain are fully parametric equalizers. The control functions are Gain (boost/cut), Frequency and Bandwidth. The Gain control is adjustable from -20 to +15 dB in 1/2 dB steps. Frequency is adjustable for the full audio range. Bandwidth is specified per octave and is adjustable from 3.0 to 0.004 octaves. Figure 8 illustrates the panel view and Figures 9a, 9b and 9c show response curves for a range of parametric sweeps.


Input Channel 1

Parametric EQ1

Gain 0.0dB

Freq 200Hz

BW 1.000

Figure 8. Panel view for one of two parametric equalizers.

LOW SHELF EQ

The response of this EQ has a shelf level that is determined by the boost or cut value. The top encoder adjusts the Gain (boost/cut) and has a range of –15 dB to +15 dB. The corner frequency is adjustable up to 200 Hz. The corner frequency is positioned 3 dB below the shelf level with a 6 dB/oct slope between the shelf point and 0 dB. Figures 10a and 10b show the panel view and frequency response sweeps for several values of gain.

COMPRESSOR

The compressor is the last process in the input chain. The output of this process feeds the matrix for the associated channel. The compressor has a total of five functions available. They are: Threshold, Ratio, Gain, Attack and Decay. Refer to figures 11a, 11b, and 11c for panel views and performance graphs.

THRESHOLD: The threshold is adjusted via the top encoder. The range is 0 to –100 dBFS in 1/2 dB steps. This control determines when the compressor begins to limit the dynamic range of the input signal. The signal strength measured at the input to the compressor is constantly monitored. The compressor has no effect upon the signal as long as the measured signal strength is below the threshold point. Once the signal strength reaches and surpasses the threshold point, the compressor will begin limiting the dynamic range of the output. The amount of dynamic compression is determined by the Ratio control.

RATIO: The Ratio of the compression is adjusted using the middle encoder. The range of control is from 1 to 20. Though it is not specified in the panel view, the ratio is referenced to 1 dB. Therefore a Ratio setting of eight is an 8 to 1 ratio. This particular setting for example means that as the input signal increases its level by 8 dB, the compressor allows only 1 dB of increase at its output.

GAIN: The output gain of the compressor is adjusted via the bottom encoder. The range of this control is –100 to +30 dB in 1/2 dB steps. This adjustment can serve to compensate for the loss in average signal strength due to the dynamic compression. If the user desires to optimally set the gain in order to maintain digital full scale and therefore maximum dynamic range, the following formula can be used:

$$Gain_{dB} = \left|Threshold_{dB}\right| * \left(1 - \frac{1}{Ratio}\right)$$

For example, given a threshold of –20 dB and a ratio of 4:1, the calculation is:

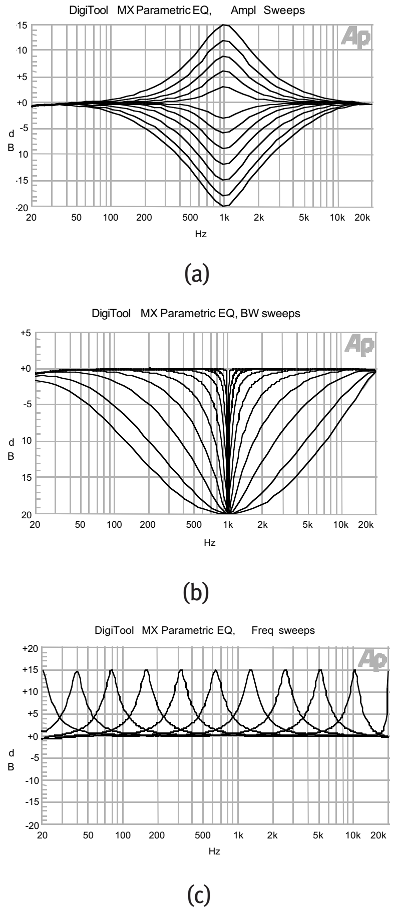
$$Gain_{dB} = 20 * \left(1 - \frac{1}{4}\right) = 15_{dB}$$

Therefore the above gain setting will result in the output level reaching maximum full scale just as the input signal reaches full scale. Note that this formula is given only to provide a mechanism to maximize dynamic range. It is not intended to compensate for average signal loss due to the compression.

ATTACK & DECAY: The Attack and Decay functions are accessed via the PROGRAM WHEEL. Attack sets the amount of time for the compressor to respond once the threshold has been met. Decay determines the amount of time for the compressor to recover from a compressed state once the input signal strength falls below the threshold. The adjustment range for each control is 1 to 1000 ms.

Edit View - Output Channel

Editing of an output channel object is identical to editing an input process. The following signal processes are:

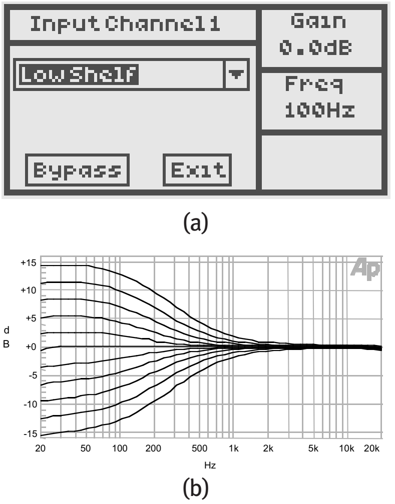


(a)

(b)

(c)

Figures 9a, 9b and 9c. Frequency response curves for a range of parametric sweeps.



Input Channel 1

Gain 0.0dB

Low Shelf

Freq 100Hz

(a)

(b)

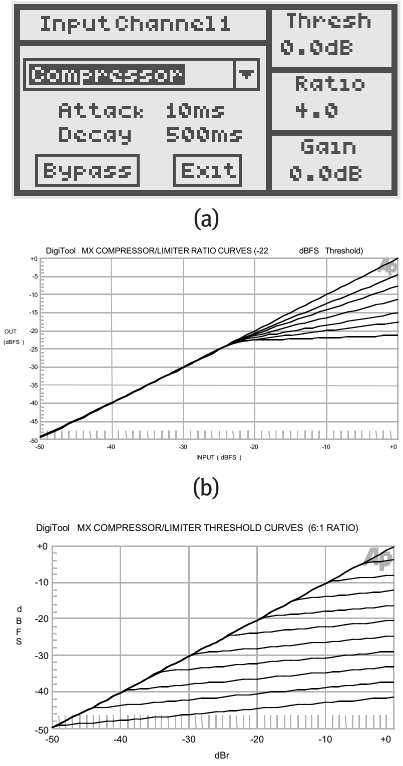
Figures 10a and 10b. Low shelving EQ panel view (a) and boost/cut response curves.

5-BAND PARAGRAPHIC

There is a five-band paragraphic equalizer for each output channel. Functionally, each band operates identically to the parametrics of the input stage. Paragraphics are named as such to differentiate their control facility from a graphic equalizer, which has fixed bandwidth and frequency. From the Digitool™ front panel, they are presented as parametric equalizers with their frequencies centered on five octaves. They are 40, 160, 640, 2.56k and 10.25 kHz, each with 1 octave bandwidths. Figure 12 displays the boost/cut response curves. For detailed operation, refer to the parametric EQ description in the "EDIT VIEW-Input Channel" section.

COMPRESSOR

Refer to the "EDIT VIEW – Input Channel" section for a description of the operation of the compressor.


Input Channel 1

Compressor

Attack 10ms

Decay 500ms

Thresh 0.0dB

Ratio 4.0

Gain 0.0dB

Figures 11a, 11b and 11c. Panel view for the compressor parameter controls.

OUTPUT CHANNEL FILTERS 1 & 2

Each output channel provides two general purpose filters for low-pass, band-pass or high-pass functions. Each filter is a 2nd order hi-pass or low-pass function (12 dB/oct. slope). A 24 dB/oct slope is possible by configuring both filters to be either low-pass or hi-pass types. Configuring both filters for a band-pass response (filter 1 = HP, fiter 2 = LP), will result in response that is limited to a 12 dB/octave slope. Each filter operates identically to the input-section filter. Refer to the "EDIT VIEW – Input channel" section for a description of the operation of the LPF-HPF filter function.

OUTPUT CHANNEL DELAY

Each output channel provides an alignment delay for up to 5 seconds. The delay is specified in milliseconds and µseconds and can be resolved to samples periods. Figure 13 provides the panel view for the delay control.

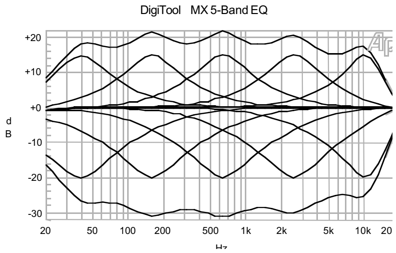


Figure 12. Response curves for the 5-band paragraphic EQ.

Utility View

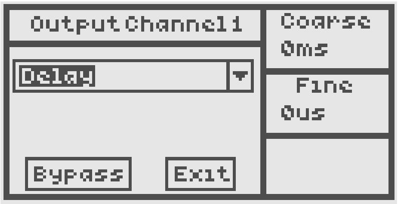
Selecting 'UTIL' in the upper left-hand corner of the Mix View panel will enable the Utility View panel. The Utility View allows the user to store and save presets, enable the internal generator and turn on the priority function.

SAVE PRESET

Parameters for a given configuration can be saved into 1 of 8 preset locations. All hardware and software variables are saved. Figure 14 illustrates the Panel View for the save function.

LOAD PRESET

Parameters for a given configuration can be reloaded from 1 of 8 previously saved preset locations. All hardware and software variables are reloaded. Figure 15 illustrates the panel view for the save function.


Output Channel 1

Delay

Coarse 0ms

Fine 0us

Figure 13. Panel view for the output delay controls.

The user also has the option to reload the factory default configuration. This configuration has all functions bypassed and each input channel routed to it's corresponding output.

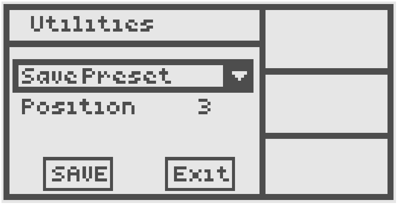


Figure 14. Panel view for the utility save function.

SIGNAL GENERATOR

An internal signal generator may be accessed from the Utility View. The generator output signal is positioned prior to the paragraphic equalizers in each of the output stages. The signal can be routed to one or more output channels. This signal, when enabled, is mixed with the normal audio signal flow for each channel. The generator provides three functions - Sine, Pink Noise and White Noise. The frequency range of the sine wave generator is 10 to 20 kHz. The attenuation range is -127 to 0 dBFS. Refer to Figure 16 for the panel view.

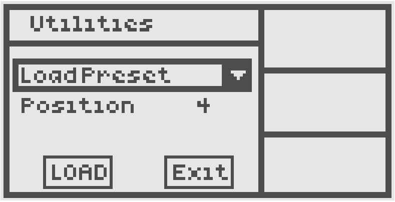


Figure 15. Panel view for the utility load function.

PRIORITY OVERRIDE

Channel one provides a priority override of channels two through eight. The Priority View provides the user with the ability to enable and select any or all channels for override (Figure 17). The Priority function is implemented via the channel one gate. When the channel one gate threshold is reached and Priority is enabled, the selected override channels will be attenuated by an envelope set by the channel one gate attack, decay and floor settings. That is, if the channel one gate attack is set for 10 ms, then slave channels will attenuate with the same attack time. Similarly, the decay and floor settings for channel one will determine the attenuation floor and decay recovery of the slave channels. The Priority Override acts on the gates of the slave channels. The gates for each slave channel still operate normally as gates until the Priority condition becomes true. In addition, the Priority Override acts on the slave channel gates even if the gate function is disabled (assuming that priority override is enabled for that channel).

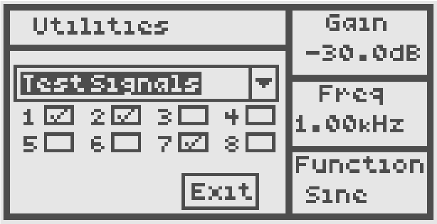


Figure 16. Panel view for the utility signal generator.

UPGRADING DIGITOOL

DigiTool firmware and DSP code can be upgraded using a Flash utility which is available from the Peavey web site. The Flash utility is a Windows-based application that allows the user to connect to the Digitool via the back-panel serial port, using a standard PC running Win95, 98 or NT. The user must select the proper comm port channel number in the utility's 'Options' pull-down menu. Once the proper communications port is selected, you can open a source file for download from the PC to Digitool. The download process takes approximately 4 minutes. At the end of the download, the Flash utility will report a successful download and instruct the user to cycle power for the new program to take affect.

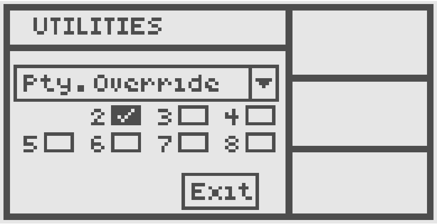


Figure 17 Channel 1 priority override view.

CAUTION: Once the download is in progress, it should not be interrupted. Digitool usually detects bad data or communication errors. Under some circumstances Digitool may attempt to load a corrupt file and become unresponsive. To recover from this condition, the user must load the upgrade using the hardware method, described below.

LOADING FIRMWARE VIA HARDWARE RESET

To download new firmware and DSP code from the PC to Digitool using the hardware method, the user must use a terminal emulator. Hyperterminal is a terminal emulation program provided by Windows which is intended for connection to a modem. It is often found in the Accessories folder under Communications. Hyperterminal should be setup as follows:

Connection name:	Digitool
Area code:	Not applicable for this connection, but you must provide one anyway
Connect using:	Com1, Com2, Com3, etc., (Select the PC communications port that is connected to Digitool)

Port Settings:

Bits per second (baudrate):	57.6k
Data bits:	8
Parity:	None
Stop bits:	1
Flow control:	None

Once you have a hyperterminal window, open the 'File' pull-down menu and select 'Properties'. Select the 'Settings' tab and change the emulation type to 'VT 100'. Select save to keep this configuration.

Next, turn the Digitool unit off and remove the top cover.

Locate the 'Flash' pads which is between U1705 and U1708 (see figure 18). The steps are as follows:

1. While the unit is still off, short across the two Flash pads using a small flat head screwdriver.
2. While still shorting the pads, turn the unit on.
3. Digitool will respond with all front panel LEDs brightly lit Green. If Digitool does not respond this way, repeat steps 1 and 2.
4. Once the LEDs are all green you may release the short.
5. In the Hyperterminal 'Transfer' pull-down menu, select 'Send Text file..'
6. Browse to the directory containing the upgrade file, select the file and press 'Open'.
7. Digitool will report 'GDone' to the emulator screen if the download was successful. If not, the message 'BDone' will appear, and therefore the process will need to be repeated.
8. If successful, cycle power on the unit to enable the new program.

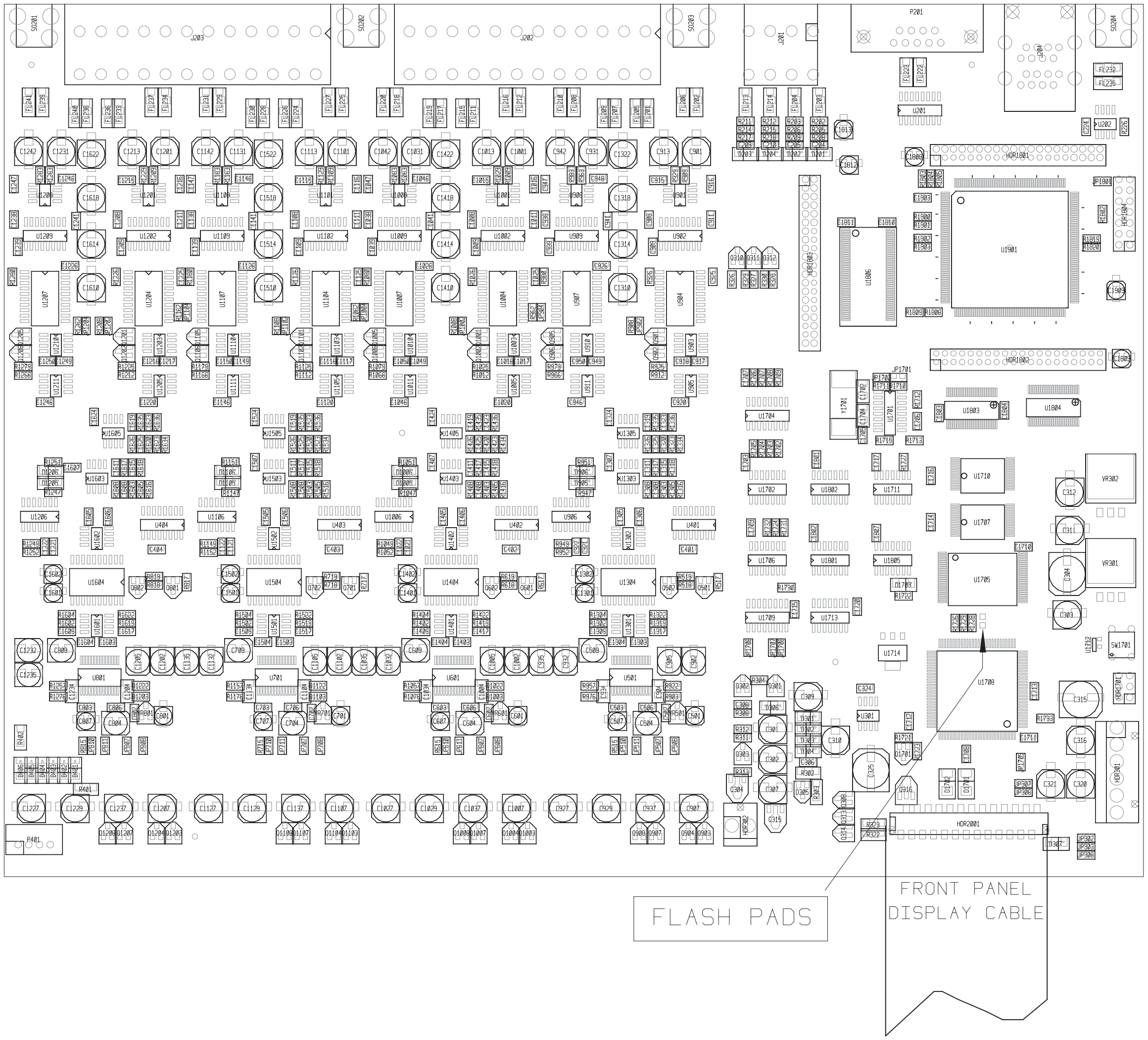
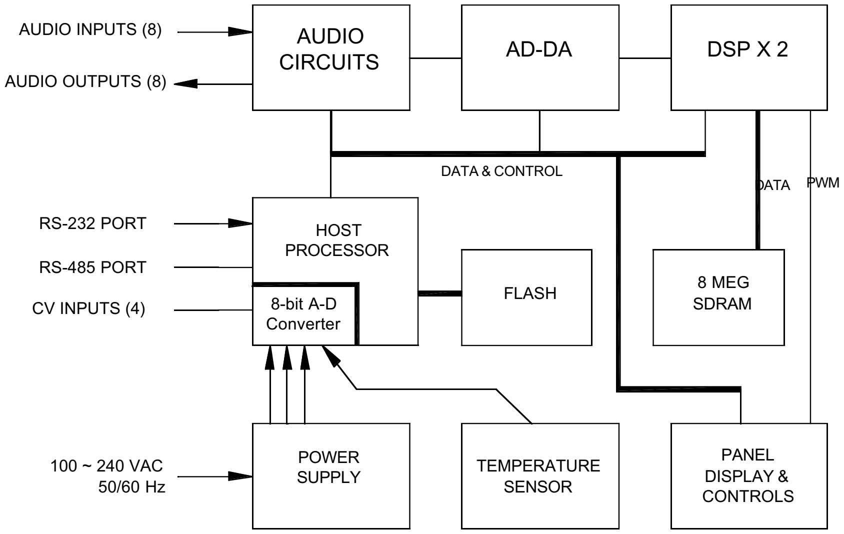
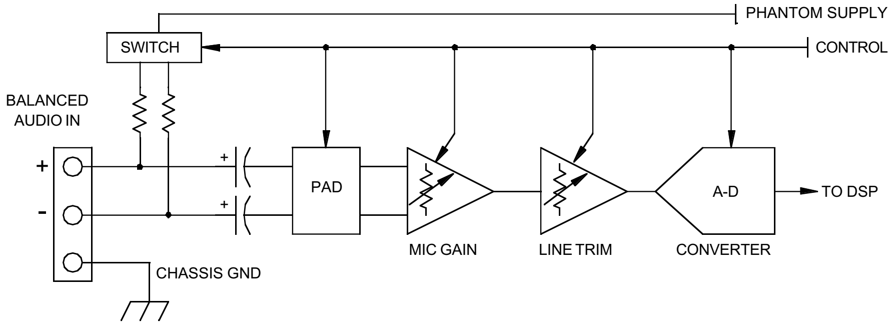

FLASH PADS
FRONT PANEL DISPLAY CABLE

Figure 18 Location of shorting pads for hardware-reset download operation.

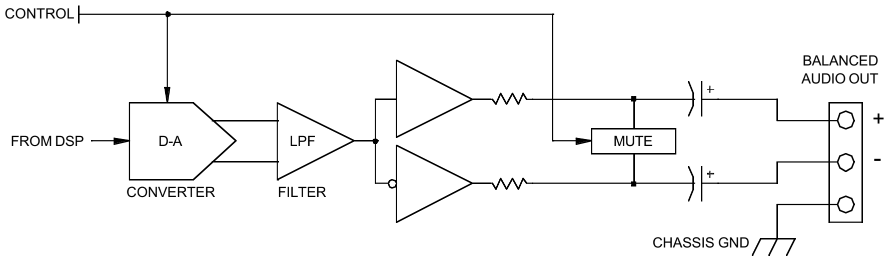
DIGITOOL MX SYSTEM LEVEL BLOCK DIAGRAM


AUDIO INPUTS (8)
AUDIO OUTPUTS (8)
AUDIO CIRCUITS
AD-DA
DSP X 2
DATA & CONTROL
DATA
PWM
RS-232 PORT
RS-485 PORT
CV INPUTS (4)
HOST PROCESSOR
8-bit A-D Converter
FLASH
8 MEG SDRAM
100 ~ 240 VAC 50/60 Hz
POWER SUPPLY
TEMPERATURE SENSOR
PANEL DISPLAY & CONTROLS

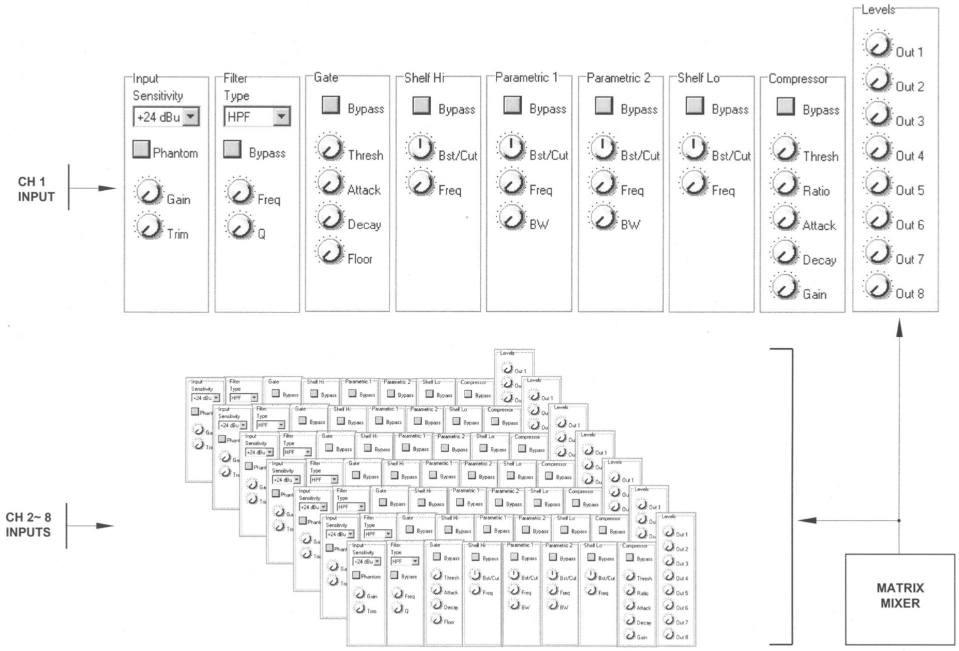
DIGITOOL MX ANALOG INPUT SIGNAL PATH - EACH CHANNEL


PHANTOM SUPPLY
SWITCH
CONTROL
BALANCED AUDIO IN
+
-
PAD
A-D
TO DSP
MIC GAIN
LINE TRIM
CONVERTER
CHASSIS GND

DIGITOOL MX ANALOG OUTPUT SIGNAL PATH - EACH CHANNEL

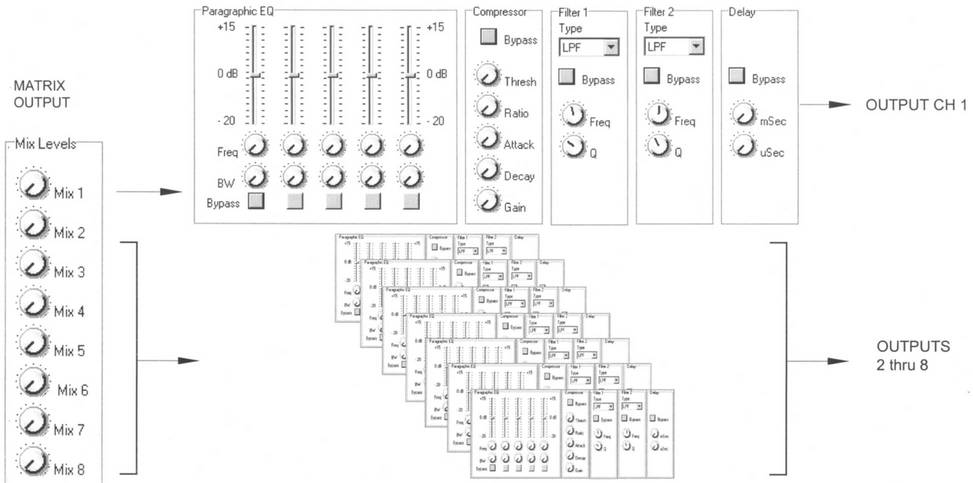

CONTROL
BALANCED AUDIO OUT
FROM DSP
D-A
LPF
MUTE
+
-
CONVERTER
FILTER
CHASSIS GND

DIGITOOL MX INPUT & DSP SIGNAL PATH (Symbolic)

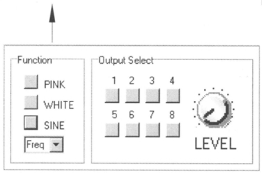

CH 1
INPUT
Input
Sensitivity
+24 dBu
Phantom
Gain
Trim
Filter
Type
HPF
Bypass
Freq
Q
Gate
Bypass
Thresh
Attack
Decay
Floor
Shelf Hi
Bypass
Bst/Cut
Freq
Parametric 1
Bypass
Bst/Cut
Freq
BW
Parametric 2
Bypass
Bst/Cut
Freq
BW
Shelf Lo
Bypass
Bst/Cut
Freq
Compressor
Bypass
Thresh
Ratio
Attack
Decay
Gain
Levels
Out 1
Out 2
Out 3
Out 4
Out 5
Out 6
Out 7
Out 8
CH 2~ 8
INPUTS
MATRIX
MIXER

Channel 1 processing is duplicated for channels 2 to 8

DIGITOOL MX DSP OUTPUT SIGNAL PATH (Symbolic)


MATRIX
OUTPUT
Mix Levels
Mix 1
Mix 2
Mix 3
Mix 4
Mix 5
Mix 6
Mix 7
Mix 8
Paragraphic EQ
+15
0 dB
- 20
+15
0 dB
- 20
Freq
BW
Bypass
Compressor
Bypass
Thresh
Ratio
Attack
Decay
Gain
Filter 1
Type
LPF
Bypass
Freq
Q
Filter 2
Type
LPF
Bypass
Freq
Q
Delay
Bypass
mSec
uSec
OUTPUT CH 1
OUTPUTS
2 thru 8

The processing shown for Mix 1 is duplicated for each output mix


Function
PINK
WHITE
SINE
Freq
Output Select
1 2 3 4
5 6 7 8
LEVEL

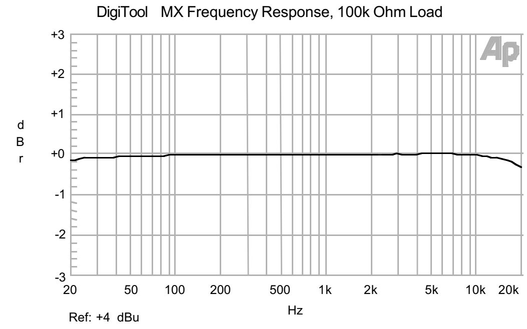
The function generator output is applied to any combination of output mixes.

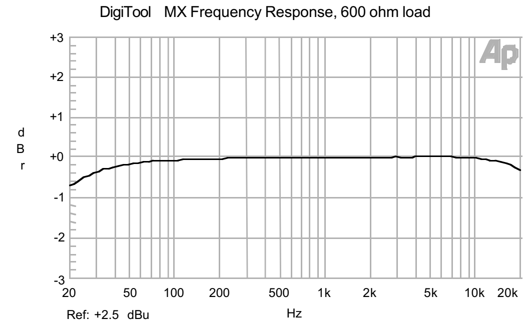
SPECIFICATIONS	MEASURE	COMMENTS
AUDIO CHANNELS	8 inputs/8 outputs	Line or Mic level inputs
LED METERING	8 input / 8 output peak-reading Dual color LEDs	LED STATUS: Red = -0.5 dBFS and higher (Clip) Red & Green = -4 to -0.5 dBFS Green = -4 to -40 dBFS Off = below -40 dBFS
INPUT		
ANALOG TRIM RANGE	-6 to +6 dB	Software controlled
LINE SENSITIVITY SETTINGS	+30 dBu, +24 dBu, +18 dBu, +12 dBu	Software selectable The sensitivity setting determines the maximum input signal that results in a full-scale digital input (0 dBFS).
MIC GAIN RANGE	+15 to +50 dB	1/2 dB steps
INPUT IMPEDANCE	6.6 k Ω	Electronically balanced
PHANTOM POWER	+24 Volts	
CMRR	55 dB	
CROSSTALK	>90 dB	@ 1 kHz
DYNAMIC RANGE	104 dB	A-weighted, 150 Ohm input termination
OUTPUT		
OUTPUT FULL SCALE SETTINGS	+24 dBu, +18 dBu, +12 dBu, 6 dBu	Software selectable The sensitivity setting determines the maximum output signal that results from a full-scale digital signal (0 dBFS).
OUTPUT IMPEDANCE	102 Ω	Electronically balanced
MINIMUM LOAD IMPEDANCE	600 Ω	Maximum output @ 600Ω = 22.5 dBu
DYNAMIC RANGE	109 dB	A-weighted
ANALOG INPUT-TO-OUTPUT		
TOTAL HARMONIC DISTORTION (+ NOISE)	0.01 %	20 Hz to 20 kHz, 22 kHz BW filter +4 dBu signal with 20 dB headroom
FREQUENCY RESPONSE	20 Hz to 20 kHz	± 0.5 dBr referenced at 1 kHz
DYNAMIC RANGE	103 dB	A-weighted, 150 Ohm input termination
LATENCY	1.4 mS	Propagation delay from input to output
DIGITAL		
SAMPLE RATE	44.1 kHz, 48 kHz, 96 kHz	
QUANTIZATION	24-bit	256 x-over-sampled Delta-sigma Analog-to-Digital and Digital-to-Analog
DIGITAL PROCESSING	132 Mips 48-bit accumulators	Uses two ADI SHARC processors
MASTER CPU CLOCK SPEED	6 MHz	
DSP CYCLE SPEED	66 MHz	
CONTROL		
RS-485	half-duplex 57.6 kbaud	1 port with 2 multi-drop RJ-45 connections
RS-232	56.7 kbaud	Standard dB-9 serial cable connector (female)
CV Inputs	0 to 10 V	Operates with either an external voltage applied or using a remote passive potentiometer connection
GENERAL		
DIMENSIONS	19" (48.26 cm) W x 12.25" (31.12 cm) D x 3.5" (8.89 cm) H	without connectors
NET WEIGHT	11.6 lbs. (5.26 kg)	
AC POWER INPUT VOLTAGE	100 VAC to 240 VAC	47 Hz to 63 Hz Universal Power Supply
POWER CONSUMPTION	20.7 Watts @100 V	330 mA
	20.5 Watts @120 V	321 mA
	22.7 Watts @240 V	206 mA
POWER DISSIPATION	70.7 BTU (17.7 kcal)	@100 V
	70.0 BTU (17.6 kcal)	@120 V
	77.5 BTU (19.5 kcal)	@240 V
AGENCY COMPLIANCE LISTINGS	UL, CUL, CE, FCC part 15 Class B	Pending

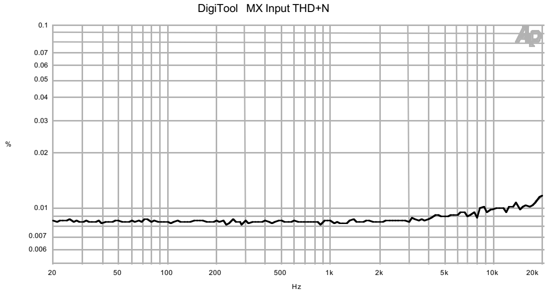
Notes:

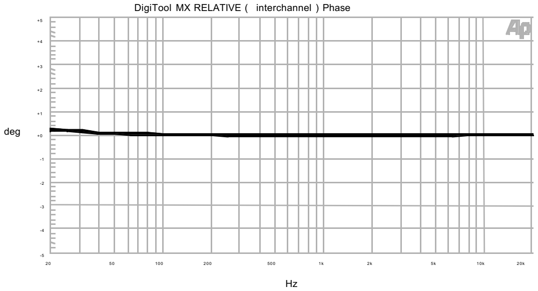
1. All specifications are typical for any channel(s).
2. All measurements are made in the digital domain.
3. All specifications are for an AC line input of 120 Volts RMS.
4. All input measurements are made using a 40 Ω balanced source impedance at +24 dBu full scale unless otherwise noted.
5. All measurements are made with gain/attenuation set for unity unless otherwise noted.

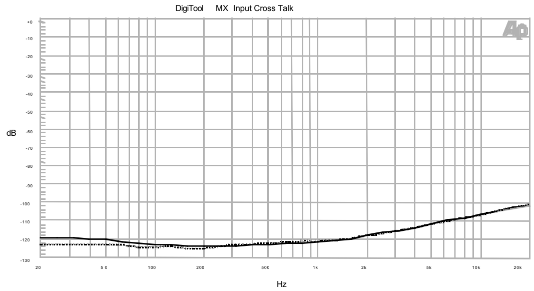
Performance Graphs

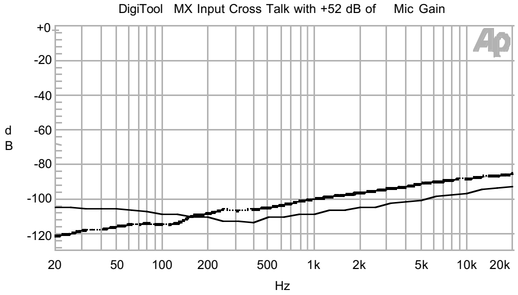


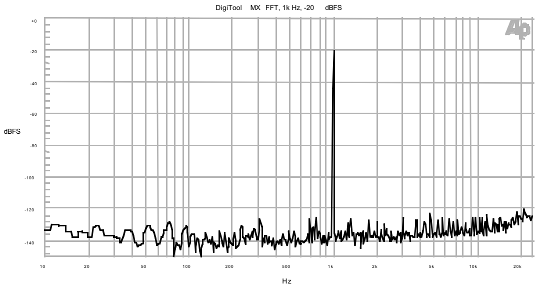












Architectural Acoustics®

PEAVEY ELECTRONICS CORPORATION LIMITED WARRANTY

Effective Date: July 1, 1998

What This Warranty Covers
Your Peavey Warranty covers defects in material and workmanship in Peavey products purchased and serviced in the U.S.A. and Canada.

What This Warranty Does Not Cover
The Warranty does not cover: (1) damage caused by accident, misuse, abuse, improper installation or operation, rental, product modification or neglect; (2) damage occurring during shipment; (3) damage caused by repair or service performed by persons not authorized by Peavey; (4) products on which the serial number has been altered, defaced or removed; (5) products not purchased from an Authorized Peavey Dealer.

Who This Warranty Protects
This Warranty protects only the original retail purchaser of the product.

How Long This Warranty Lasts
The Warranty begins on the date of purchase by the original retail purchaser. The duration of the Warranty is as follows:

Product Category	*Duration*
MediaMatrix® DPU (Excluding Frames), Cinema Processors, Power Amplifiers, Pre-Amplifiers, Mixers, Electronic Crossovers and Equalizers	5 years
Loudspeakers	5 years
Microphones	2 years
Frames	1 year
Speaker Components (incl. speakers, baskets, drivers, diaphragm replacement kits and passive crossovers) and all Accessories	1 year

What Peavey Will Do
We will repair or replace (at Peavey's discretion) products covered by warranty at no charge for labor or materials. If the product or component must be shipped to Peavey for warranty service, the consumer must pay initial shipping charges. If the repairs are covered by warranty, Peavey will pay the return shipping charges.

How To Get Warranty Service
(1) Take the defective item and your sales receipt or other proof of date of purchase to your Authorized Peavey Dealer or Authorized Peavey Service Center.
OR
(2) Ship the defective item, prepaid, to Peavey Electronics Corporation, International Service Center, 412 Highway 11 & 80 East, Meridian, MS 39301 or Peavey Canada Ltd., 95 Shields Court, Markham, Ontario, Canada L3R 9T5. Include a detailed description of the problem, together with a copy of your sales receipt or other proof of date of purchase as evidence of warranty coverage. Also provide a complete return address.
OR
(3) All MediaMatrix® Frames needing repair, should be shipped prepaid to Peavey Electronics Corporation, International Service Center, 412 Highway 11 & 80 East, Meridian, MS 39301

Limitation of Implied Warranties
ANY IMPLIED WARRANTIES, INCLUDING WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE, ARE LIMITED IN DURATION TO THE LENGTH OF THIS WARRANTY.
Some states do not allow limitations on how long an implied warranty lasts, so the above limitation may not apply to you.

Exclusions of Damages
PEAVEY'S LIABILITY FOR ANY DEFECTIVE PRODUCT IS LIMITED TO THE REPAIR OR REPLACEMENT OF THE PRODUCT, AT PEAVEY'S OPTION. IF WE ELECT TO REPLACE THE PRODUCT, THE REPLACEMENT MAY BE A RECONDITIONED UNIT. PEAVEY SHALL NOT BE LIABLE FOR DAMAGES BASED ON INCONVENIENCE, LOSS OF USE, LOST PROFITS, LOST SAVINGS, DAMAGE TO ANY OTHER EQUIPMENT OR OTHER ITEMS AT THE SITE OF USE, OR ANY OTHER DAMAGES WHETHER INCIDENTAL, CONSEQUENTIAL OR OTHERWISE, EVEN IF PEAVEY HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.
Some states do not allow the exclusion or limitation of incidental or consequential damages, so the above limitation or exclusion may not apply to you.

This Warranty gives you specific legal rights, and you may also have other rights which vary from state to state.

If you have any questions about this warranty or service received or if you need assistance in locating an Authorized Service Center, please contact the Peavey International Service Center at (601) 483-5365 / Peavey Canada Ltd. at (905) 475-2578.

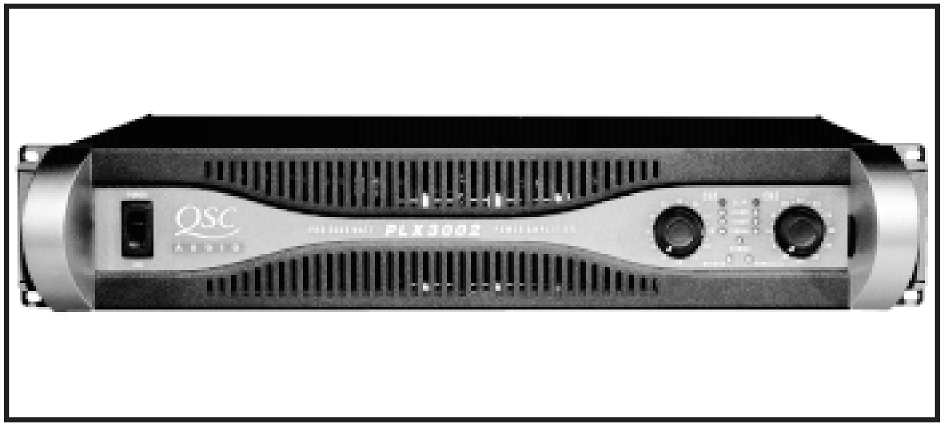
Features and specifications subject to change without notice.

80303050

Peavey Electronics Corporation • 711 A Street • Meridian, MS 39301
601-483-5376 • Fax 601-486-1678 • http://aa.peavey.com
© 2001
Printed in the U.S.A. 11/01

Architectural Acoustics™ By Peavey
Listen to This™

QSC™
AUDIO



▼ PLX 1202

▼ PLX 1602

▼ PLX 2402

▼ PLX 3002

▼ PLX 3402

User Manual

Manuel de l’utilisateur

Bedienhandbuch

Manual del Usuario

PLX™ Series Amplifiers

TD-000072-00
Rev. C

Explanation of graphical symbols

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated “dangerous voltage” within the product’s enclosure that may be of sufficient magnitude to constitute a risk of electric shock to humans.

The exclamation point within an equilateral triangle is intended to alert the users to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: To reduce the risk of electric shock, do not remove the cover. No user-serviceable parts inside. Refer servicing to qualified service personnel.

WARNING: To prevent fire or electric shock, do not expose this equipment to rain or moisture.

Explication des symboles graphiques

Le symbole éclair avec pointe de flèche à l'intérieur d'un triangle équilatéral est utilisé pour alerter l'utilisateur de la présence à l'intérieur du coffret de "tension non-isolée dangereuse" d'ampleur suffisante pour constituer un risque de choc électrique pour l'être humain.

Le point d'exclamation à l'intérieur d'un triangle équilatéral est employé pour alerter les utilisateur de la présence d'instructions importantes pour le fonctionnement et l'entretien (service) dans les documents accompagnant l'appareil.

ATTENTION!

RISQUE DE CHOC
ÉLECTRIQUE
NE PAS OUVRIR

ATTENTION: Pour éviter les risques de choc électrique, ne pas enlever le couvercle. Cet appareil ne comporte aucune pièce pouvant être réparée par l'utilisateur. Confier l'entretien à un technicien qualifié.

AVERTISSEMENT: Pour éviter le risque de choc électrique ou d'incendie, n'exposez cet appareil ni à l'humidité excessive ni aux projections d'eau (pluie, ruissellement, etc ...)

Erklärung der Bildsymbole

Das Blitzzeichen innerhalb eines gleichseitigen Dreiecks warnt den Benutzer vor nicht isolierter, gefährlicher Spannung im Inneren des Gerätes. Diese Spannung ist hoch genug, um Personen durch elektrischen Schlag zu gefährden.

Das Ausrufungszeichen innerhalb eines gleichseitigen Dreiecks weist den Benutzer auf wichtige Bedienungs- und Wartungsanweisungen hin, die in den gerätebegleitenden Unterlagen aufgeführt sind.

VORSICHT

GEFAHR EINES
ELEKTRISCHEN SCHLAGES.
NICHT ÖFFNEN!

VORSICHT: Um Gefährdung durch elektrischen Schlag zu vermeiden, darf das Gehäuse nicht geöffnet werden. Es befinden sich keine vom Benutzer reparierbaren Teile im Inneren des Gerätes. Überlassen Sie jegliche Reparatur dem qualifizierten Fachmann.

WARNUNG: Um die Gefahr eines Brandes bzw. eine Verletzung durch elektrischen Schlag zu vermeiden, sollten Sie das Gerät niemals Regen oder Feuchtigkeit aussetzen.

Explicación de símbolos

El rayo dentro de un tríangulo equilátero alerta al usuario de la presencia de voltaje peligroso no aislado dentro del producto, que puede tener un nivel suficiente para constitutuir un riesgo de descarga eléctrica para el usario.

El signo de exclamación inscrito en un triángulo equilátero alerta a los usuarios de la presencia de instrucciones importantes de funcionamiento y mantenimiento (servicio) en la literatura que acompaña al producto.

PRECAUCIÓN

RIESGO DE DESCARGA
ELÉCTRICA. NO LO ABRA.

PRECAUCIÓN: Para reducir el riesgo de descarga eléctrica, no quite la tapa. El usario no debe ajustar los componentes internos. Para mantenimiento solicite la ayuda de personal cualificado.

AVISO: Para evitar un incendio o una descarga eléctrica, no exponga este equipo a la lluvia o humedad.

TABLE OF CONTENTS ▼ TABLE DES MATIÈRES ▼ INHALTSVERZEICHNIS ▼ TABLA DE LAS MATERIAS

- **Explanation of graphical symbols** 2
  - *Explication des symboles graphiques*
  - *Erklärung der Bildsymbole*
  - *Explicación de símbolos*
- **TABLE OF CONTENTS** 3
  - *Table des matières*
  - *Inhaltsverzeichnis*
  - *Tabla de las materias*
- **Declaration of Conformity** 4
  - *Déclaration de Conformité*
  - *Konformitätserklärung*
  - *Declaración de Conformidad*
- **INTRODUCTION** 5–7
  - *AVANT-PROPOS*
  - *EINFÜHRUNG*
  - *INTRODUCCIÓN*
- **Front panel** 6
  - *Panneau avant*
  - *Vorderseite*
  - *Panel frontal*
- **Rear panel** 7
  - *Panneau arrière*
  - *Rückseite*
  - *Panel posterior*
- **FEATURES & SETUP** 8–13
  - *CARACTÉRISTIQUES ET LEUR UTILISATION*
  - *AUSSTATTUNG & EINSTELLUNGEN*
  - *CARACTERÍSTICAS Y AJUSTES*
- **Clip limiter** 8
  - *Limiteur d'écrêtement*
  - *Clip Limiter*
  - *Limitador de picos*
- **Input filter** 9
  - *Filtres d'entrée*
  - *Eingangsfilter*
  - *Filtro de entrada*
- **Parallel input mode** 10
  - *Mode entrées parallèles*
  - *Eingangsparallelschaltung*
  - *Modo de entradas paralelas*
- **Bridge mono mode** 11
  - *Mode ponté mono*
  - *Monobrückenbetrieb*
  - *Modo puenteado en mono*
- **What are the differences among Stereo, Parallel Input, and Bridge Mono modes?** 12
  - *Modes stéréo, parallèle et ponté, quelles sont les différences?*
  - *Unterschiede zwischen Stereo-, Parallel- und Monobrückenbetrieb*
  - *¿Cuáles son las diferencias entre los modos Estéreo, Entradas Paralelas y Puenteado en Mono?*
- **INSTALLATION** 14
  - *INSTALLATION*
  - *EINBAU*
  - *INSTALACIÓN*
- **CONNECTIONS** 15–17
  - *CONNEXIONS*
  - *ANSCHLÜSSE*
  - *CONEXIONES*
- **Inputs** 15
  - *Entrées*
  - *Eingänge*
  - *Entradas*
- **Speakon™ outputs** 16
  - *Sorties Speakon™*
  - *Speakon™ Ausgänge*
  - *Salidas Speakon™*
- **Binding post outputs** 17
  - *Bornes à écrou*
  - *Anschlussklemmen*
  - *Terminales con tornillo*
- **Operating voltage (AC mains)** 17
  - *Tension d'utilisation (alimentation CA)*
  - *Netzanschluß*
  - *Voltaje de operación (CA principal)*
- **OPERATION** 18, 19
  - *UTILISATION*
  - *BETRIEB*
  - *OPERACIÓN*
- **AC power switch** 18
  - *Interrupteur d'alimentation CA*
  - *Netzschalter*
  - *Interruptor de encendido*
- **Gain controls** 18
  - *Contrôles de gain*
  - *Verstärkungsregler*
  - *Controles de ganancia*
- **LED indicators** 18
  - *Indicateurs DEL*
  - *LED-Anzeige*
  - *Indicadores LED*
- **Fan cooling** 19
  - *Ventilation*
  - *Lüfterkühlung*
  - *Ventilación*
- **Safe operating levels** 19
  - *Niveaux d'utilisation sécuritaires*
  - *Betriebspegel*
  - *Niveles adecuados de operación*
- **APPLICATIONS** 20, 21
  - *APPLICATIONS*
  - *ANWENDUNGSBEISPIELE*
  - *APLICACIONES*
- **Sound reinforcement** 20
  - *Sonorisation*
  - *Beschallung*
  - *Sonido en vivo*
- **Instrument amplification** 21
  - *Amplification d'instrument*
  - *Instrument-Verstärkung*
  - *Amplificación de instrumento*
- **TROUBLESHOOTING** 22–25
  - *DÉPANNAGE*
  - *FEHLERBEHEBUNG*
  - *BUSQUEDA DE AVERÍAS*
- ***Problem:* no sound** 22
  - *Problème:* pas de son
  - *Problem:* kein Ton
  - *Problema:* no hay sonido
- ***Problem:* distorted sound** 23
  - *Problème:* son distortionné
  - *Problem:* Verzerrungen
  - *Problema:* sonido distorsionado
- ***Problem:* no channel separation** 24
  - *Problème:* pas de séparation des canaux
  - *Problem:* keine Kanaltrennung
  - *Problema:* no hay separación entre los canales
- ***Problem:* hum** 24
  - *Problème:* ronflement
  - *Problem:* Brummen
  - *Problema:* zumbidos
- ***Problem:* hiss** 25
  - *Problème:* sifflement
  - *Problem:* Rauschen/Zischen
  - *Problema:* ruido (*hiss*)
- ***Problem:* squeals and feedback** 25
  - *Problème:* bruits et effet Larsen
  - *Problem:* Pfeifen und Rückkopplungen
  - *Problema:* chillidos y retroalimentación
- **INNER WORKINGS** 26, 27
  - *SOUS LE CAPOT*
  - *INNERER AUFBAU*
  - *OPERACIÓN INTERNO*
- **SPECIFICATIONS** 28, 29
  - *SPÉCIFICATIONS*
  - *TECHNISCHE DATEN*
  - *ESPECIFICACIONES*
- **WARRANTY INFORMATION** 30
  - *INFORMATIONS DE GARANTIE*
  - *GARANTIE-BEDINGUNGEN*
  - *INFORMACIÓN DE GARANTÍA*
- **ADDRESS & TELEPHONE INFORMATION** 31
  - *ADRESSE POSTALE ET NUMÉROS*
  - *ADRESSE UND TELEFONNUMMERN*
  - *DIRECCIÓN Y TELÉFONO*

CE 98

NOTE:

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- **Reorient or relocate the receiving antenna.**
- **Increase the separation between the equipment and the receiver.**
- **Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.**
- **Consult the dealer or an experienced radio/TV technician for help.**

Declaration of Conformity

QSC Audio Products, Inc. declares as its sole responsibility that the above named product is in compliance with the Council Directive 89/336/EEC of 3 May 1989 on the harmonization of the laws of the Member States relating to electromagnetic compatibility, as well as requirements of the harmonized product standard EN 55013 related to the limits and the methods of measurement of radio disturbance characteristics of broadcast receivers and associated equipment, and EN 55020, related to the electromagnetic immunity of broadcast receivers and associated equipment.

QSC Audio Products, Inc. further declares as its sole responsibility that the above named product is in compliance with the Council Directive 73/23/EEC of 19 February 1973 on the harmonization of the laws of the Member States relating to electrical equipment for use within certain voltage limits and the Council Directive 93/68/EEC of 22 July 1993 which amends the previous mentioned directive, as well as requirements of the harmonized product standard EN 60065, related to safety requirements of the harmonized product standard EN 60065, related to safety requirements for mains operated electronic and related apparatus for household and similar general use.

Déclaration de Conformité

QSC Audio Products Inc déclare être seul responsable que le produit nommé ci-dessus est en conformité avec la directive européenne 89/336/EEC du 3 mai 1989 concernant l’harmonisation des lois des états membres relatives à la compatibilité électromagnétique, ainsi que de les spécifications de la norme harmonisée EN 55013 relative aux limites et méthodes de mesure des perturbations radio des récepteurs de télédiffusion et équipements associés, et EN 55020 relative à l’immunité électromagnétique des récepteurs de télédiffusion et équipements associés.

QSC Audio Products Inc déclare de plus être seul responsable que le produit nommé ci-dessus est en conformité avec la directive européenne 73/23/EEC du 19 février 1973 concernant l’harmonisation des lois des états membres relatives aux équipements électriques pour un usage dans une certaine limite de tension, et à la directive 93/68/EEC du 22 juillet 1993 qui amende la directive précédente, ainsi qu’aux spécifications de la norme harmonisée EN 60065, relative à la sécurité pour les appareils électriques d’usage domestique et général connectés au réseau électrique.

Konformitätserklärung

QSC Audio Products, Inc. erklärt in alleiniger Verantwortung, daß dieses Produkt der Ratsdirektive 89/336/EEC vom 3. Mai 1989 zur Gesetzesharmonisierung der Mitgliedstaaten betreffend der elektromagnetischen Kompatibilität entspricht, sowie die Erfordernisse der Einheitlichen Norm EN55013 bezüglich Grenzen und Messmethoden von Rundfunkstörungscharakteristik von Rundfunkempfängern und zugehörigen Geräten erfüllt, sowie der Einheitlichen Norm EN55020 bezüglich der elektromagnetischen Strahlungssicherheit von Rundfunkempfängern und zubehörigen Geräten.

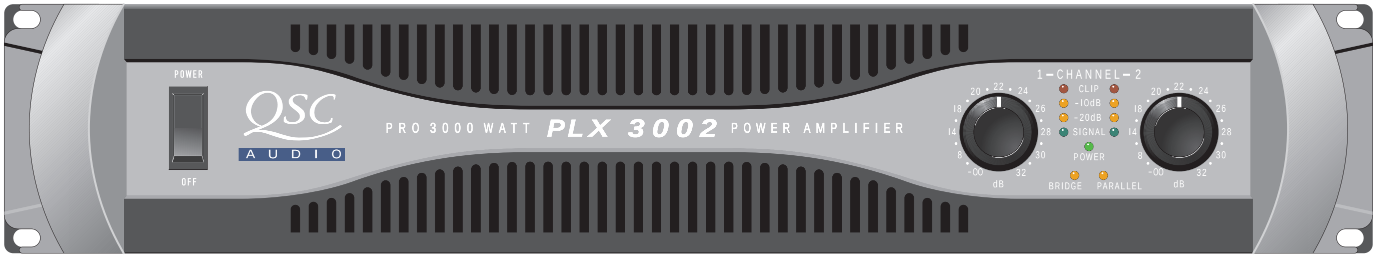
QSC Audio Products, Inc. erklärt ferner, daß das oben bezeichnete Produkt der Ratsdirektive 72/23/EEC vom 19. Februar 1973 entspricht, in dem die Gesetzesharmonisierung der Mitgliedstaaten bezogen auf elektrische Geräte zur Benutzung innerhalb bestimmter Spannungsgrenzen festgelegt sind, sowie der Ratsdirektive 93/68/EEC vom 22. Juli 1993 mit Änderungsbestimmungen, sowie der Einheitlichen Produktnorm EN60065 entspricht, in der Sicherheitsbestimmungen für netzbetriebene elektronische und ähnliche Geräte für Haushaltsgebrauch oder ähnlichen generellen Gebrauch festgelegt sind.

Declaración de Conformidad

QSC Audio Products, Inc. declara bajo su propia responsabilidad que el producto anteriormente mencionado cumple con el Concejo Directivo 89/336/EEC del 3 de Mayo de 1989 en la armonización de las leyes de los Estados Miembros que se relacionan a la compatibilidad electromagnética, como también a los requerimientos del producto armonizado con la norma EN 55013 relacionados a los límites y métodos de medidas de perturbaciones de radio que son características de los receptores de la radiodifusión y equipo asociado, y la EN 55020 relacionada con la inmunidad electromagnética de receptores de radiodifusión y equipo asociado.

QSC Audio Products, Inc. además declara bajo su propia responsabilidad que el producto ya mencionado cumple con el Concejo Directivo 72/23/EEC del 19 de Febrero de 1973 en la armonización de la leyes de los Estados Miembros relacionados al equipo eléctrico para uso dentro de ciertos límites de voltaje y el Concejo Directivo 93/68/EEC del 22 de Julio de 1993 el cual reforma la directiva previamente mencionada, como también los requerimientos de la norma del producto armonizado EN 60065, relacionado a los requerimientos de seguridad de los circuitos eléctricos que se operan en aparatos electrónicos relacionados para el hogar y similarmente para uso general.

INTRODUCTION | AVANT-PROPOS | EINFÜHRUNG | INTRODUCCIÓN



Model	Power, 8Ω/ch 20 Hz–20 kHz, 0.03% THD	Power, 4Ω/ch 20 Hz–20 kHz, 0.05% THD	Power, 2Ω/ch 1 kHz, 1% THD
PLX 1202	200 W	325 W	600 W
PLX 1602	300 W	500 W	800 W
PLX 2402	425 W	700 W	1200 W
PLX 3002	550 W	900 W	1500 W
PLX 3402	700 W	1100 W	1700 W

© Copyright 1998 & 1999, QSC Audio Products, Inc.

QSC® is a registered trademark, and PowerWave™ is a trademark, of QSC Audio Products, Inc.

"QSC" and the QSC logo are registered with the U.S. Patent and Trademark Office.

All other trademarks are the property of their respective owners.

The PLX Series Amplifiers. PowerWave™ technology and a refined audio structure combine studio-quality performance with touring-quality portability. These fan-cooled, 2-channel, 2-RU, amps provide unprecedented power in a strong, compact chassis.

Features

- Independent, user-defeatable clip limiters
- Fully selectable low-frequency filtering; choice of 30 or 50 Hz roll-off
- Stereo (dual-channel), parallel-input, or bridged mono operating modes
- Balanced inputs, both XLR and ¼" (6.3 mm) TRS
- Binding post and Neutrik Speakon™ outputs
- Detented gain controls
- Front panel LED indicators for power, signal, -20 and -10 dB, clip/protect, parallel inputs, and bridged mono mode
- QSC's exclusive high-performance PowerWave switching technology power supply

Les amplificateurs de la série PLX. La technologie PowerWave™ et une circuiterie audio rafinée sont combinées pour obtenir une qualité sonore digne d'un studio d'enregistrement, avec la robustesse requise pour la tournée. Ces amplificateurs ventilés, 2 canaux, occupant 2 espaces normalisés, vous procurent un niveau de puissance sans précédent dans un châssis compact et solide.

Caractéristiques

- Limiteurs d'écrêtement indépendants, commutables par l'utilisateur
- Filtres passe-haut commutables, avec choix de coupure à 30 ou 50 Hz
- Modes d'opération stéréo (deux canaux), parallèle, ou ponté mono
- Entrées symétriques sur prises XLR et ¼" (6.3 mm)
- Sorties sur bornes à écrou et prises Speakon
- Contrôles de gain à détentes
- Indicateurs à DEL sur le panneau avant pour alimentation, présence de signal, -20 dB, -10 dB, écrêtement/protection, modes mono ponté et parallèle
- Bloc d'alimentation à commutation haute performance utilisant le technologie PowerWave exclusive à QSC

Die PLX Verstärkerserie. Die Verbindung einer hochentwickelten Audioverarbeitung mit der patentierten PowerWave™ Technologie kombiniert studiotaugliche Tonqualität mit den robusten Eigenschaften einer Touring-Endstufe. Diese kompakten, lüftergekühlten 2-Kanal-Endstufen mit 2 HE liefern unübertroffen hohe Leistung in einem stabilen Gehäuse.

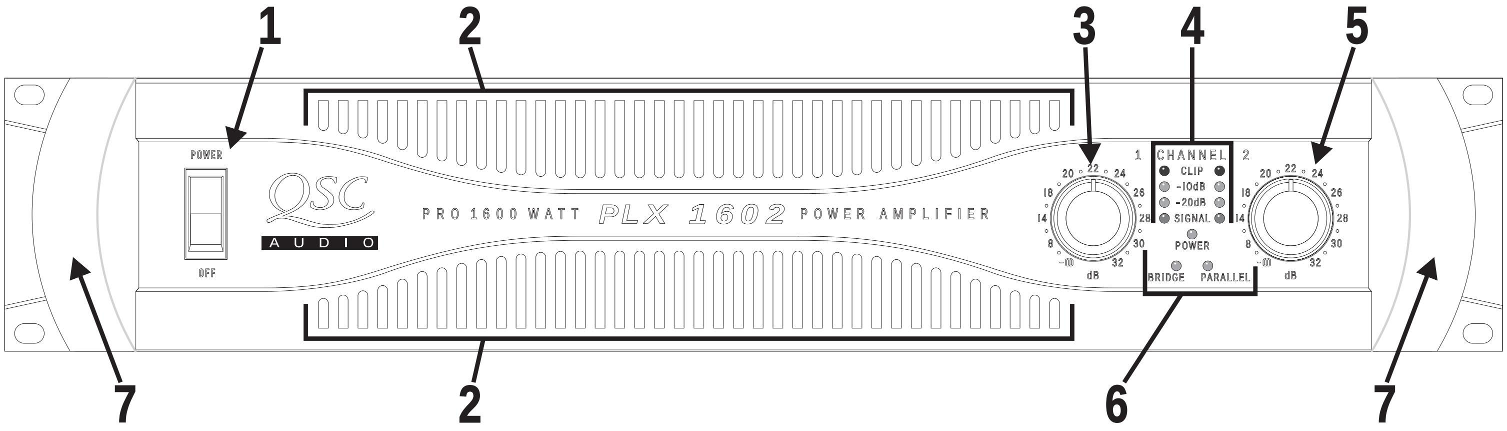
Ausstattungsmerkmale

- unabhängige, abschaltbare Spitzenbegrenzer
- Einstellbare Hochpassfilter: 30 oder 50 Hz Eckfrequenz
- Stereo (Zweikanal)-, Parallel- oder Mono Brückenbetrieb
- Symmetrische Eingänge—XLR und Stereoklinke
- Neutrik Speakons und Ausgangsklemmen
- Rasterpotis
- LED Frontanzeigen für: Power, -20 dB, -10dB und Clip/Protect, parallelen und Monobrückenbetrieb
- QSC's patentierte PowerWave Netzteile

La línea de amplificadores PLX. La tecnología PowerWave™ y una refinada estructura sonora se han combinado para ofrecerle la calidad de estudio con la portabilidad necesaria para el sonido en vivo. Estos amplificadores de dos canales con enfriamiento por ventiladores y que ocupan dos espacios en un rack, ofrecen una increíble potencia en un compacto y resistente armazón.

Características

- Limitador de picos independiente
- Filtro de frecuencias graves completamente seleccionables; con la elección de atenuación entre 30 ó 50 Hz
- Estéreo (dos canales), modos de operación para entradas paralelas, o "puenteadas" en mono
- Entradas balanceadas, con conectores XLR y ¼" (6.3mm) TRS
- Postes de amarre y salidas Neutrik Speakon™
- Controles de ganancia con retén
- Indicadores LED en el panel frontal para el encendido, la señal de entrada, niveles de -20 y -10 dB, para protección/saturación, entradas paralelas, modo "puenteado" en mono
- Tecnología de encendido exclusiva de QSC PowerWave en la fuente de alimentación


1
2
3
4
5
POWER
OFF
QSC AUDIO
PRO 1600 WATT PLX 1602 POWER AMPLIFIER
1 CHANNEL 2
CLIP
-10dB
-20dB
SIGNAL
POWER
BRIDGE
PARALLEL
dB
7
2
6
7

Front panel

1. Power switch
2. Cooling vents
3. Gain control (Channel 1)
4. ***CLIP***, ***-10 dB***, ***-20 dB*** and ***SIGNAL*** indicator LEDs, both channels
5. Gain control (Channel 2)
6. ***POWER***, ***BRIDGE***, and ***PARALLEL*** indicator LEDs
7. Handles

Panneau avant

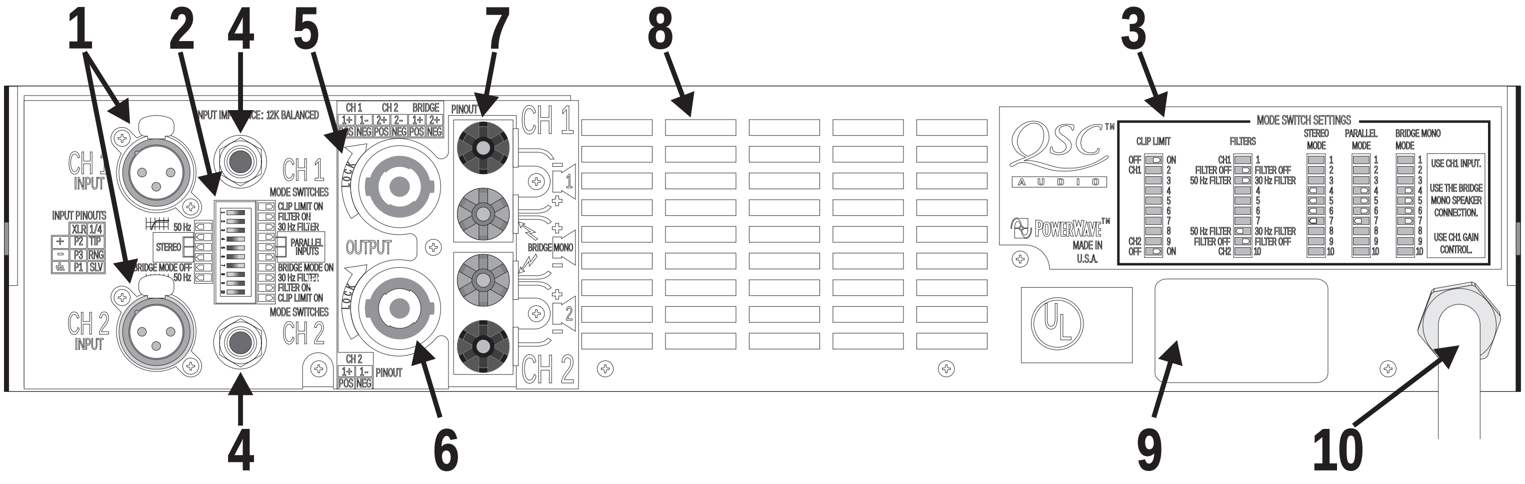
1. Commande marche/arrêt
2. Bouches de ventilation
3. Commande de gain (Canal 1)
4. DELs ***CLIP*** (écrêtage), ***-10 dB***, ***-20 dB***, et ***SIGNAL*** (tous les deux canaux)
5. Commande de gain (Canal 2)
6. DELs ***POWER*** (alimentation), ***BRIDGE*** (mode mono ponté), et ***PARALLEL*** (entrées parallèles)
7. Poignées

Vorderseite

1. Netzschalter
2. Abluftöffnungen
3. Pegelsteller (Kanal 1)
4. LED-Anzeige für ***CLIP***, ***-10 dB***, ***-20 dB*** und ***SIGNAL*** (beide Kanäle)
5. Pegelsteller (Kanal 2)
6. LED-Anzeige für ***POWER*** (Betrieb), ***BRIDGE*** (Mono-Brückenschaltung) und ***PARALLEL*** (Eingangsparallelschaltung)
7. Griffe

Panel frontal

1. Interruptor de encendido
2. Rejillas de ventilación
3. Control de ganancia (Canal 1)
4. Indicadores LED de ***CLIP***, ***-10 dB***, ***-20 dB*** y ***SIGNAL*** (dos canales)
5. Control de ganancia (Canal 2)
6. Indicadores LED de ***POWER*** (encendido), ***BRIDGE*** (puenteado en mono) y ***PARALLEL*** (entradas paralelas)
7. Asas


1
2
4
5
7
8
3
INPUT IMPEDANCE: 12K BALANCED
CH 1 INPUT
CH 2 INPUT
INPUT PINOUTS
XLR 1/4
+ P2 TIP
− P3 RNG
P1 SLV
50 Hz
STEREO
BRIDGE MODE OFF
50 Hz
CH 1 MODE SWITCHES
CLIP LIMIT ON
FILTER ON
30 Hz FILTER
PARALLEL INPUTS
BRIDGE MODE ON
30 Hz FILTER
FILTER ON
CLIP LIMIT ON
MODE SWITCHES CH 2
CH 1 CH 2 BRIDGE
1+ 1- 2+ 2- 1+ 2+
POS NEG POS NEG POS NEG
LOCK
OUTPUT
CH 2
1+ 1-
POS NEG
PINOUT
CH 1
BRIDGE MONO
CH 2
QSC AUDIO
POWERWAVE
MADE IN U.S.A.
MODE SWITCH SETTINGS
CLIP LIMIT
FILTERS
STEREO MODE
PARALLEL MODE
BRIDGE MONO MODE
OFF CH1 ON
CH1 FILTER OFF FILTER OFF
50 Hz FILTER 30 Hz FILTER
50 Hz FILTER 30 Hz FILTER
FILTER OFF FILTER OFF CH2
CH2 OFF ON
USE CH1 INPUT.
USE THE BRIDGE MONO SPEAKER CONNECTION.
USE CH1 GAIN CONTROL.
UL
4
6
9
10

Rear panel

1. XLR inputs, Channels 1 and 2
2. Configuration switch
3. Configuration switch chart
4. TRS inputs, Channels 1 and 2
5. Speakon output, Channel 1 plus Channel 2
6. Speakon output, Channel 2
7. Binding post outputs, Channels 1 and 2
8. Cooling air inlet vents
9. Serial number label
10. AC power cable

Panneau arrière

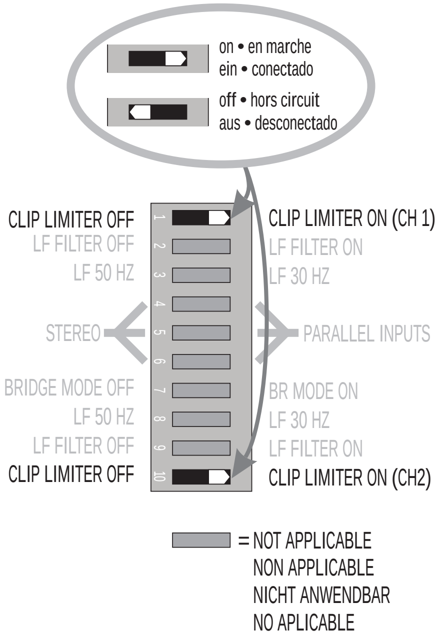
1. Entrées XLR (Canaux 1 et 2)
2. Sélecteur de configuration
3. Diagramme au sélecteur de configuration
4. Entrées TRS (Canaux 1 et 2)
5. Sortie Speakon (Canal 1 plus Canal 2)
6. Sortie Speakon (Canal 2)
7. Sorties sur bornes à écrou (Canaux 1 et 2)
8. Bouches de ventilation
9. Étiquette du numéro en série
10. Câble d’alimentation secteur

Rückseite

1. XLR-Eingänge (Kanäle 1 und 2)
2. Konfigurationsschalter
3. Erläuterung des Konfigurationsschalters
4. Stereoklinken (Kanäle 1 und 2)
5. Speakon-Ausgang (Kanal 1 mit Kanal 2)
6. Speakon-Ausgang (Kanal 2)
7. Ausgangsklemmer (Kanäle 1 und 2)
8. Lufteinlaßöffnung
9. Seriennummer
10. Netzkabel

Panel posterior

1. Entradas XLR (Canales 1 y 2)
2. Selector de configuración
3. Esquema del selector de configuración
4. Entradas TRS (Canales 1 y 2)
5. Salida Speakon (Canal 1 con Canal 2)
6. Salida Speakon (Canal 2)
7. Salidas postes de amarre
8. Rejillas de ventilación
9. Etiqueta del número de serie
10. Cable de alimentación



FEATURES & SETUP

Clip limiter

WHAT IT IS

When the audio signal drives the amp's output circuit beyond its power capability, it clips, flattening the peaks of the waveform. The clip limiter detects this and quickly reduces the gain to minimize the amount of overdrive. To preserve as much of the program dynamics as possible, limiting occurs only during actual clipping.

Each channel has its own clip limiter, and you can switch it on or off independently, as shown at left.

WHEN TO USE IT (OR NOT)

When driving full-range speakers, clip limiting reduces high frequency distortion caused by bass overloads. It also protects higher frequency drivers from excess overdrive and harsh clipping harmonics.

When driving subwoofers, some users let the amplifier clip without limiting because it gives extra “punch” to kick drums and similar sounds.

CAUTION: Clip limiting reduces extreme overdrive peaks, allowing a higher average signal level without audible distortion. However, increasing the gain with the clip limiter engaged, until clipping is again audible, can double the average output power. Be careful not to exceed the power rating of your speakers.

CARACTÉRISTIQUES ET LEUR UTILISATION

Limiteur d'écrêtement

DESCRIPTION

Quand le signal audio dépasse la capacité de puissance du circuit de sortie de l'amplificateur, le signal est écrêté, tronquant les pointes de la forme d'onde. Le circuit limiteur d'écrêtement détecte cette condition et réduit rapidement le gain de l'amplificateur pour minimiser la surtension. Afin de préserver le maximum de la dynamique du signal, le limiteur ne reste en fonction que pour la durée de l'écrêtement.

Chaque canal est muni de son propre limiteur, que vous pouvez activer et désactiver indépendamment, tel qu'illustré à gauche.

UTILISATION

Lors de l'utilisation de l'amplificateur avec un signal plein registre, le circuit limiteur réduit la distortion des hautes fréquences causée par les surtensions des basses fréquences. Le circuit protège aussi les haut-parleurs de hautes fréquences des surtensions et des harmoniques dues aux écrêtements.

En utilisation avec des haut-parleurs de sous-graves, certains utilisateurs préfèrent laisser le circuit limiteur hors-circuit, on peut ainsi obtenir un son avec plus d'impact dans la reproduction des grosses caisses et autres instruments du même type.

ATTENTION; le circuit de limiteur d'écrêtement réduit les pointes de surtension, augmentant le niveau moyen du signal sans distortion audible. On peut ainsi facilement doubler la puissance moyenne livrée aux haut-parleurs. Il faut prendre soin de ne pas excéder la puissance admissible des haut-parleurs.

AUSSTATTUNG & EINSTELLUNGEN

Clip Limiter

BESCHREIBUNG

Wenn das Audiosignal die Ausgangsverstärker übersteuert, wird das Signal geklippt, was zu einer Abflachung der Wellenspitzen führt. Der Clip Limiter (Spitzenbegrenzer) erkennt dies und verringert schnell die Verstärkung um die Stärke der Übersteuerung zu reduzieren. Um dabei gleichzeitig aber soviel Dynamik als möglich zu erhalten, spricht der Spitzenbegrenzer nur während echter Übersteuerungen an.

Jeder Kanal hat seinen eigenen Limiter, der wahlweise zu- oder abgeschaltet werden kann (siehe Zeichnung).

EINSATZ DES LIMITERS

Bei der Verwendung von Breitband-Lautsprechersystemen reduziert der Einsatz eines Spitzenbegrenzers die Hochfrequenzverzerrungen, welche von Übersteuerungen im Bassbereich verursacht werden. Außerdem werden die Höhentreiber vor Übersteuerung und hart geklippten Harmonischen geschützt.

Einige Anwender bevorzugen aber die Übersteuerung des Verstärkers um bei Sub-Woofer Systemen zusätzlich Punch für die Bass-Drum oder ähnlich perkussive Instrumente zu erhalten.

ACHTUNG: Der Einsatz eines Spitzenbegrenzers ermöglicht eine höheres Durchschnittssignal ohne hörbare Verzerrungen. Eine Erhöhung der Verstärkung mit eingeschaltetem Clip Limiter kann die durchschnittliche Ausgangsleistung auf das Doppelte erhöhen. Bitte achten Sie daher darauf, die Leistungsangaben Ihrer Lautsprecher nicht zu überschreiten.

CARACTERÍSTICAS Y AJUSTES

Limitador de picos

¿QUÉ ES?

Cuando la señal de audio estimula el circuito de salida del amplificador más allá de su capacidad de potencia se satura con los picos de la forma de onda. El limitador de picos detecta este fenómeno y rápidamente reduce la ganancia para minimizar la cantidad de sobresaturación. Para conservar las dinámicas del programa al máximo, la limitación sólo ocurrira durante el instante mismo de la saturación.

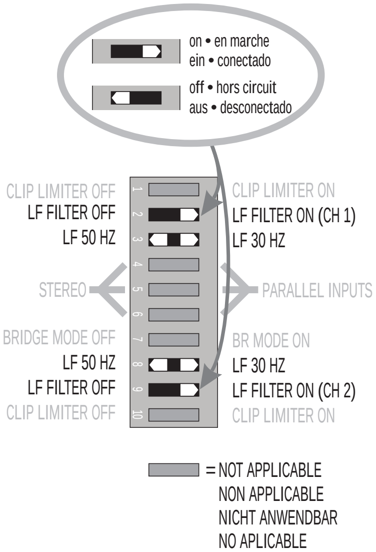
Cada canal tiene su propio limitador de picos, usted los puede activar o desactivar independientemente, como se muestra a la izquierda.

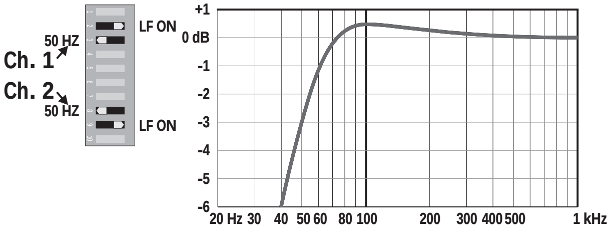
¿CUÁNDO (O CUÁNDO NO) USARLO?

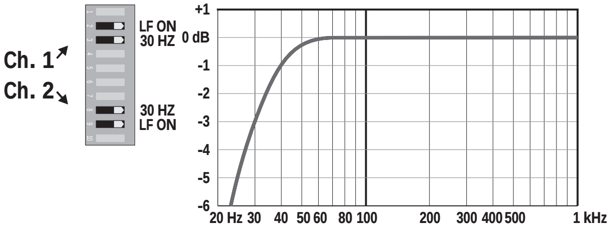
Cuando se usan bocinas de rango completo, el limitador de picos reduce la distorsión de las frecuencias agudas causada por la sobrecarga de graves. También protege a los drivers de frecuencias agudas del exceso de saturación y armónicos descontrolados.

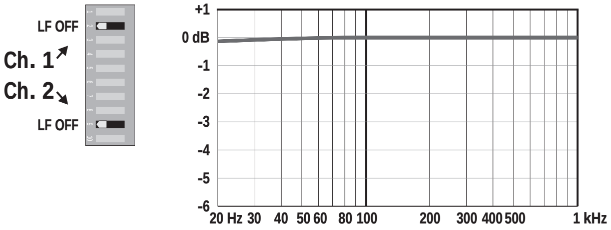
Cuando se usan subwoofers, algunos prefieren usar los picos del amplificador sin limitación para obtener un "punch" adicional en el bombo de la batería y otros sonidos similares.

ADVERTENCIA: El limitador de picos reduce señales extremas de saturación, permitiendo un nivel de señal más alto sin distorsión audible. Sin embargo, aumentar la ganancia con el limitador de picos activado puede doblar la capacidad de salida. Tenga cuidado de no rebasar el límite de potencia de sus bocinas.









FEATURES & SETUP

Input filter

WHAT IT IS

The low-frequency (LF) filter rolls off signals below either 30 Hz or 50 Hz. This improves bass performance by limiting sub-audio cone motion, making more power available for the speakers' rated frequency range.

The filter settings for each channel are controlled individually through the DIP switch settings shown. When the filter is turned off, a 5 Hz rolloff protects against DC or deep sub-audio inputs.

WHEN TO USE IT (OR NOT)

As a rule, your speakers will sound better with proper filtering. Unless you already have filtering in a preceding device, match the setting to the low frequency rating of your speakers. Vented (bass reflex, ported, etc.) speakers are especially sensitive to cone over-excursion at frequencies below their rated limit.

The 50 Hz filter works well with most compact full-range speakers, and has a slight boost at 100 Hz for greater fullness. The 30 Hz filter is intended for subwoofers and large full-range cabinets. The "off" position should be used only for applications such as studio playback monitoring, where you need to know if there are unwanted sub-audio signals present in your mix.

CARACTÉRISTIQUES ET LEUR UTILISATION

Filtres d'entrée

DESCRIPTION

Le filtre passe-haut, lorsqu'activé, réduit le niveau de signal sous 30 Hz ou 50 Hz. Ce filtre améliore la performance du système en limitant le mouvement excessif du cône du haut-parleur de basses à des fréquences infrasoniques, donnant plus de puissance disponible dans le spectre audible.

Les ajustements des filtres pour chaque canal se font séparément sur les interrupteurs tel que montré ci-contre. Lorsque les filtres sont désactivés, un filtre passe-haut à 5 Hz entre en fonction afin de protéger le haut-parleur contre les tensions continues, et contre les signaux infrasoniques.

UTILISATION

En règle générale, les haut-parleurs fonctionnent mieux avec un filtre bien ajusté. A moins que d'autres équipements en amont dans la chaîne soient munis de filtre, ajustez le filtre de votre amplificateur en accord avec votre haut-parleur. Les enceintes à évent (bass reflex, etc) sont particulièrement sensibles aux surexcursion des cônes aux fréquences inférieures à la fréquence d'accord de l'enceinte.

Le filtre à 50 Hz fonctionne très bien avec la plupart des enceintes plein registre compactes, et le filtre augmente le gain autour de 100 Hz, pour un son plus riche. Le filtre à 30 hz est prévu pour les enceintes de sous-graves et pour les grosses enceintes plein registre. La position "off", ou dérivation, devrait être utilisée dans des applications telles l'écoute en studio, là où on doit vérifier l'absence de signal infrasonique dans le mix.

AUSSTATTUNG & EINSTELLUNGEN

Eingangsfilter

BESCHREIBUNG

Ein LF- oder Hochpassfilter schneidet Frequenzen unterhalb 30 Hz, bzw. 50 Hz ab. Hierdurch wird die Basswiedergabe verbessert, da ultratiefe, störende Frequenzen abgeschnitten werden, und hierdurch mehr Leistung für die normale Wiedergabe zur Verfügung steht.

Die Filtereinstellungen werden durch den links abgebildeten DIP-Schalter eingestellt. Bei abgeschaltetem Filter werden die Frequenzen unterhalb 5 Hz unterdrückt, damit ein Schutz von Gleichspannung und Infraschallanteilen besteht.

EINSATZ DES EINGANGSFILTERS

In der Regel werden Ihre Lautsprecher bei richtiger Filterung besser klingen. Falls nicht schon an anderer Stelle eine entsprechende Filterung stattgefunden hat, stellen Sie die Filterung entsprechend dem Frequenzgang Ihrer Lautsprecher ein. Belüftete Lautsprecher (z.B. Bassreflex, etc.) sind ausgesprochen empfindlich gegen zu große Auslenkungen der Membran unterhalb des spezifizierten Frequenzbereichs.

Der 50 Hz Filter arbeitet bestens mit praktisch allen Breitbandlautsprechern, da er zusätzlich eine leichte Verstärkung im Bereich um 100 Hz aufweist, was in einem volleren Klang resultiert. Der 30 Hz Filter ist für Subwoofer und große Breitbandkabinette gedacht. Die "Off" Einstellung sollte nur für Anwendungen wie z.B. eine Studiobeschallung verwendet werden; wo es darauf ankommt, Infraschallanteile im Mix zu erkennen.

CARACTERÍSTICAS Y AJUSTES

Filtro de entrada

¿QUÉ ES?

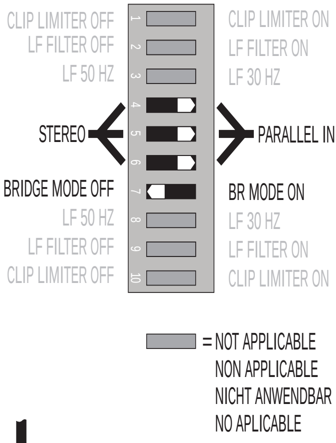
El filtro de frecuencias graves (LF) le permite atenuar señales abajo de los 30 Hz ó 50 Hz. Esto mejora el rendimiento de las frecuencias graves limitando una porción de audio no audible que mueve el cono, dejando más potencia disponible para el rango de frecuencia de las bocinas.

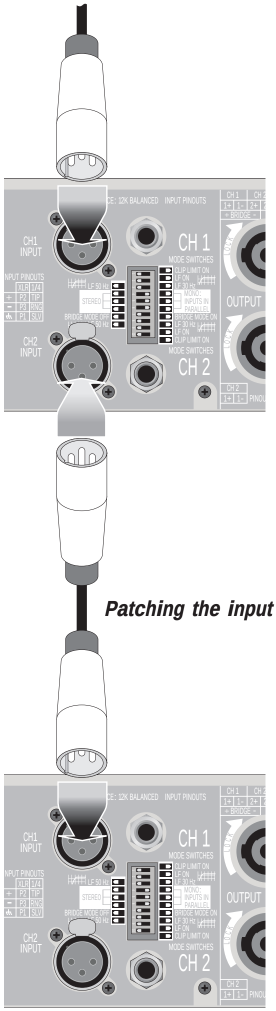
Los ajustes del filtro para cada canal se controlan individualmente con el interruptor tipo DIP, como se describe. Un atenuador de 5 Hz ofrece protección contra DC o entradas de frecuencias ultragraves cuando se apaga el filtro.

¿CUÁNDO (O CUÁNDO NO) USARLO?

Por regla, sus bocinas sonarán mejor con el filtro apropiado. A menos que anteriormente haya utilizado otro aparato para filtrar y haya igualado las frecuencias graves de sus bocinas. Las bocinas con ventilación (como la bass reflex, ported, etc …) son especialmente sensibles al desplazamiento exagerado de los conos causado por frecuencias por debajo de su límite.

El filtro de 50 Hz funciona bien con la mayoría de las bocinas de rango completo, y ofrece un suave realce a los 100 Hz para reforzar el cuerpo del sonido. El filtro de 30 Hz está diseñado para subwoofers y para gabinetes de rango completo de gran tamaño. La posición "off"—apagado—puede usarse en trabajos como monitoreo en el estudio, donde usted necesita saber si en su mezcla hay señales ultragraves no deseadas.





Patching the input signal to additional amps

FEATURES & SETUP

Parallel input mode

WHAT IT IS

The "Parallel Input" switches let you operate the amplifier in parallel mode, delivering the same signal to both channels without using a Y-cable. Each channel drives its own speaker load, with independent gain, filtering, and clip limiting.

Set switch positions 4, 5, and 6 *"ON"* to couple the inputs together. ***Turn the switches off for stereo, bi-amping, or other 2-channel modes.*** The yellow ***PARALLEL*** LED indicator on the front panel warns you when the switches are set to parallel.

With the inputs in parallel, you can use the other set of input connectors to carry the signal to other amps. This is often called a "daisy-chain."

WHEN TO USE IT

Parallel the inputs when driving two speakers with one input signal *(parallel mode)* while keeping separate control of both channels' gain, filtering, and limiting. Use them in *bridged mono mode* to patch the signal to additional amplifiers through the extra input jacks. See page 12 for an explanation of amp operating modes.

NOTE: If you're using a balanced signal, use only balanced patch cables; even one unbalanced cable will unbalance the entire signal chain, possibly causing hum.

NOTE: Turn off the "Parallel Inputs" switches when feeding the amp two separate signals.

CARACTÉRISTIQUES ET LEUR UTILISATION

Mode entrées parallèles

DESCRIPTION

Les interrupteurs "Parallel Inputs" vous permettent d'utiliser l'amplificateur en mode parallèle, où le même signal est livré aux deux canaux sans avoir à utiliser un câble "Y". Chaque canal amplifie le signal indépendamment, avec ajustements propres de gain, de filtre, et de limiteur.

Ajustez les interrupteurs 4, 5, et 6 en position "on" pour joindre les entrées des deux canaux. Laissez les interrupteurs en position "off" pour utilisation en mode stéréo, bi-amplification, ou toute autre application 2 canaux. La DEL jaune ***PARALLEL*** sur le panneau avant allume pour indiquer que l'amplificateur est réglé en mode parallèle.

Lorsque les entrées sont réglées en mode parallèle, vous pouvez utiliser le connecteur de la seconde entrée pour envoyer le signal vers un autre amplificateur. On appelle ce type branchement le chaînage en guirlande, ou "daisy chaining".

UTILISATION

Joignez les entrées (mode parallèle) quand vous voulez contrôler deux haut-parleurs avec des ajustementes de gain, de filtre et de limiteur à partir d'un même signal. Voir la page 12 pour les explications des modes d'opération.

NOTE; si vous utilisez un signal symétrique, n'utilisez que des câbles construit à cet effet, puisque l'insertion d'un seul câble asymétrique dans la chaîne rendra la chaîne asymétrique, avec tous les désavantages de ce type de connexion.

NOTE; désengager les interrupteurs "parallel inputs" quand vous amenez des signaux différents aux deux canaux.

AUSSTATTUNG & EINSTELLUNGEN

Parallelbetrieb

BESCHREIBUNG

Durch die "Parallel Input" Schalter kann der Verstärker parallel betrieben werden, wobei das gleiche Signal beiden Kanälen zugeführt wird. Jeder Verstärkerkanal treibt dabei seinen eigenen Lautsprecher mit unabhängiger Verstärkung, Filterung und Spitzenbegrenzung.

Stellen Sie die Schalterpositionen 4, 5 und 6 auf "ON" um die Eingänge zu verkoppeln. Schalten Sie die Schalter aus für Stereo-, Bi-Amping- oder einen anderen 2-Kanalbetrieb. Die gelbe ***PARALLEL***-LED auf der Frontplatte leuchtet auf, wenn Parallelbetrieb gewählt wurde.

Mit parallelgeschalteten Eingängen können die anderen Eingangsstecker benutzt werden, um das Signal an weitere Verstärker zu liefern.

EINSATZ DES PARALLELBETRIEBS

Schalten Sie die Eingänge parallel, wenn zwei Lautsprecher mit einem Signal angefahren, und dabei aber separate Verstärkung, Filterung oder Begrenzung benutzt werden soll. Wählen Sie Bridged Mono Mode (Monobrückenbetrieb) um die Signale an zusätzliche Verstärker mit Hilfe der übrigen Eingangsbuchsen zu verbinden (siehe auch Seite 12 für weitere Erläuterungen).

BITTE BEACHTEN SIE: Wenn Sie ein symmetrisches Eingangssignal anlegen, benutzen Sie auch ausschließlich symmetrische Kabel zur Weiterverbindung, da auch nur ein einziges unsymmetrisches Kabel das Gesamtsignal unsymmetrisch werden läßt, was möglicherweise Brummen verursachen kann.

BITTE BEACHTEN SIE: Schalten Sie den Parallelbetrieb aus, wenn zwei separate Signale eingespeist werden sollen.

CARACTERÍSTICAS Y AJUSTES

Modo de entradas paralelas

¿QUÉ ES?

El interruptor "Parallel Input" le permite operar el amplificador en modo paralelo, enviando la misma señal hacia ambos canales sin necesidad de utilizar un cable tipo "Y". Cada canal controla su propia carga de bocinas, con ganancia independiente, filtros y limitador de picos.

Ajuste los selectores 4, 5 y 6 en la posición de "ON", para acoplar las entradas. Coloque los interruptores en la posición de "OFF" para trabajar en el modo estéreo, biamplificado, o cualquier otro modo de 2 canales. El indicador LED amarillo para el modo ***PARALLEL*** del panel frontal, le advierte cuando el interruptor ha sido ajustado en el modo paralelo.

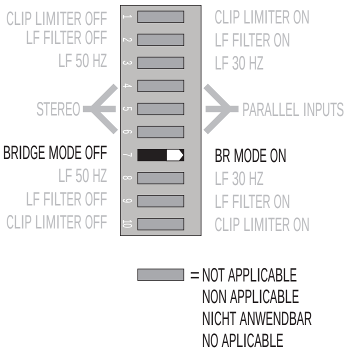
Con las entradas en paralelo, usted puede usar los conectores de la otra entrada para enviar la señal a otros amplificadores. Lo que comúnmente se llama una cadena "daisy".

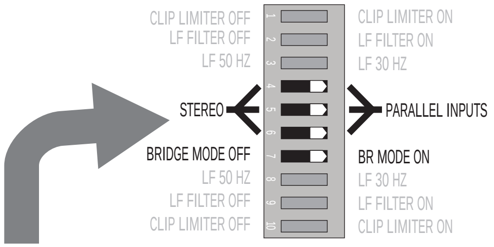
¿CUÁNDO USARLO?

Utilice la entrada paralela cuando se alimentan dos bocinas con una sola señal de entrada (modo paralelo) mientras se mantiene control por separado de la ganancia, los filtros y la limitación de ambos canales. Use el modo "puenteado" en mono para conectar la señal a amplificadores adicionales por medio de los conectores extra de entrada. Consulte la página 12 para una explicación completa de los modos de operación del amplificador.

NOTA: Si usa una señal balanceada, utilice únicamente cables adecuados. Un solo cable no balanceado puede afectar el recorrido de la señal y producir zumbidos.

NOTA: Cuando alimente el amplificador con dos señales independientes, coloque el interruptor "Parallel Inputs" en la posición de apagado—"Off".


CLIP LIMITER OFF
LF FILTER OFF
LF 50 HZ
STEREO
BRIDGE MODE OFF
LF 50 HZ
LF FILTER OFF
CLIP LIMITER OFF
CLIP LIMITER ON
LF FILTER ON
LF 30 HZ
PARALLEL INPUTS
BR MODE ON
LF 30 HZ
LF FILTER ON
CLIP LIMITER ON
= NOT APPLICABLE
NON APPLICABLE
NICHT ANWENDBAR
NO APLICABLE

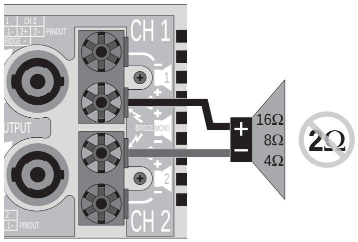

CLIP LIMITER OFF
LF FILTER OFF
LF 50 HZ
STEREO
BRIDGE MODE OFF
LF 50 HZ
LF FILTER OFF
CLIP LIMITER OFF
CLIP LIMITER ON
LF FILTER ON
LF 30 HZ
PARALLEL INPUTS
BR MODE ON
LF 30 HZ
LF FILTER ON
CLIP LIMITER ON

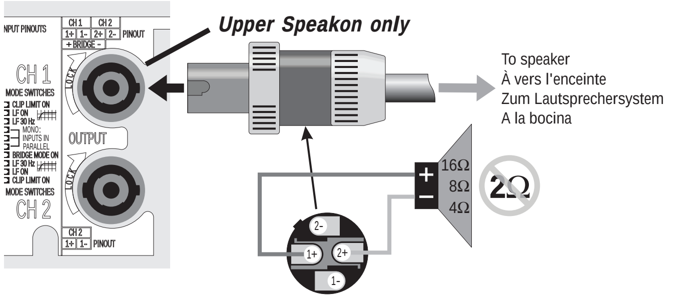
To patch the signal to additional amplifiers, use the parallel input switches described on page 10.

Pour amener le signal vers d'autres amplificateurs, suivez les instructions de la page 10.

Um das Eingangs-signal weiteren Verstärkern zur Verfügung zu stellen, verwenden Sie die Parallel Input Schalter wie auf Seite 10 beschrieben.

Para pasar la señal a los amplificadores adicionales utilice el selector de entrada paralela descrito en la página 10.


CH 1
CH 2
PINOUT
OUTPUT
BRIDGE MONO
16Ω
8Ω
4Ω
2Ω


Upper Speakon only
INPUT PINOUTS
CH 1
CH 2
PINOUT
+ BRIDGE –
MODE SWITCHES
CLIP LIMIT ON
LF ON
LF 30 Hz
MONO: INPUTS IN PARALLEL
BRIDGE MODE ON
OUTPUT
16Ω
8Ω
4Ω
2Ω
2-
1+
2+
1-
To speaker
À vers l'enceinte
Zum Lautsprechersystem
A la bocina

Bridged mono • Mono ponté • Monobrückenbetrieb • Mono puente

FEATURES & SETUP

Bridge mono mode

WHAT IT IS

Bridged mono mode combines the power of both amp channels into one speaker, resulting in twice the voltage swing, four times the peak power, and approximately three times the sustained power of a single channel. This mode uses Channel 1's input, gain control, input filter, and clip limiter; Channel 2's have no effect.

The ***BRIDGE*** LED on the front panel indicates when the amp is in bridged mono mode.

WHEN TO USE IT (OR NOT)

Use bridged mono to deliver the power of both channels to a ***single*** 8- or 4-ohm load. Set switch position 7 to "BRIDGE MONO ON." Use Channel 1's inputs, and connect the speaker as shown.

BRIDGED-MONO PRECAUTIONS:

This mode puts a high demand on the amplifier and speaker, Excessive clipping may cause protective muting or speaker damage. Be sure the speaker has a sufficient power rating.

Output voltages greater than 100 volts rms are available between the amplifier's bridged terminals. NEC CLASS 3 wiring methods, as specified in accordance with national and local codes, must be used to connect the speaker.

CARACTÉRISTIQUES ET LEUR UTILISATION

Mode ponté mono

DESCRIPTION

Le mode ponté mono combine la puissance des deux canaux pour utilisation sur un seul haut-parleur, doublant la tension, quadruplant la puissance de crête, et donnant environ le triple de la puissance continue par rapport à un canal simple. En mode ponté, utiliser l'entrée, le gain, le filtre et le limiteur du canal 1, le canal 2 étant sans effet.

La DEL ***BRIDGE*** sur le panneau avant allume pour indiquer que l'amplificateur est réglé en mode ponté mono.

UTILISATION

Utilisez le mode ponté mono pour amener la puissance des deux canaux sur une seule charge de 4 ou 8 ohms. Ajustez l'interrupteur 7 en position "on". Utilisez l'entrée du canal 1 et branchez la charge tel qu'indiqué ci-contre.

PRÉCAUTIONS EN MODE MONO PONTÉ:

Le mode ponté mono place un stress supplémentaire sur l'amplificateur et le haut-parleur. L'écrêtage excessif peut causer la mise en sourdine par le circuit de protection et/ou peut endommager le haut-parleur. Veuillez vous assurer que le haut-parleur peut accepter la puissance de l'amplificateur.

Des pointes de tension de plus de 100 volts rms sont possibles entre les bornes de sortie de l'amplificateur en mode ponté mono. Installez votre système selon les codes électriques local et national du site d'installation.

AUSSTATTUNG & EINSTELLUNGEN

Monobrückenbetrieb

BESCHREIBUNG

Der Monobrückenbetrieb addiert die Leistung beider Kanäle um sie einem Lautsprecher zur Verfügung zu stellen. Das heißt, es steht ungefähr die doppelte Ausgangsspannung, die vierfache Spitzenleistung und ungefähr die dreifache Dauerleistung eines Einzelkanals zur Verfügung. Diese Betriebsart verwendet das Eingangssignal von Kanal 1, seine Verstärkung, Filter und Begrenzung. Die Regler von Kanal 2 haben keine Wirkung.

Die ***BRIDGE*** LED auf der Frontplatte leuchtet auf, wenn diese Betriebsart gewählt wurde.

EINSATZ DES MONOBRÜCKENBETRIEBS

Verwenden Sie diese Betriebsart, um die Leistung beider Kanäle einer einzigen 8 oder 4Ω Last zur Verfügung zu stellen. Stellen Sie den Schalter 7 auf "BRIDGE MONO ON". Verwenden Sie die Eingänge von Kanal 1 und schließen Sie die Lautsprecher wie aus der Zeichnung ersichtlich an.

BRÜCKENBETRIEB VORSICHTSMAßNAHMEN:

Diese Betriebsart stellt hohe Anforderungen an Verstärker und Lautsprecher. Übermäßige Übersteuerung kann zu Abschalten des Verstärkers oder auch Lautsprecherbeschädigungen führen. Stellen Sie sicher, daß der Lautsprecher entsprechende Leistungen verarbeiten kann.

Ausgangsspannungen von mehr als 100 Volt rms liegen zwischen den Ausgangsanschlüssen an. Daher müssen die einschlägigen Sicherheitsmaßnahmen beim Anschluß der Lautsprecher beachtet werden.

CARACTERÍSTICAS Y AJUSTES

Modo puenteado en mono

¿QUÉ ES?

El modo puenteado en mono combina la potencia de ambos canales de amplificador hacia una bocina, aumentando dos veces el consumo de voltaje, cuatro veces la potencia de picos y aproximadamente tres veces la potencia de sostenimiento de un solo canal. Este modo utiliza la entrada, el control de ganancia, filtro y limitador del canal 1. Los controles del canal 2 no producen ningún efecto.

El LED marcado ***BRIDGE*** en el panel frontal, indica la posición del amplificador en el modo "puenteado" en mono.

¿CUÁNDO (O CUÁNDO NO) USARLO?

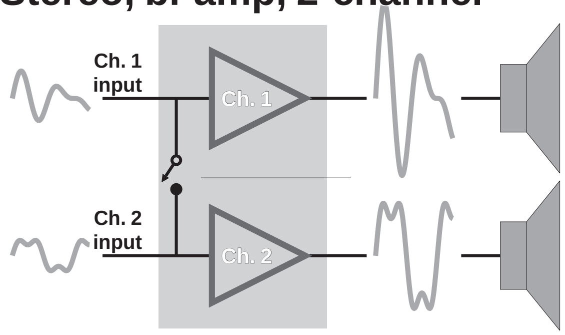
Utilice el modo puenteado en mono para pasar la potencia de ambos canales, a una sola carga de 8 ó 4 ohmios. Ajuste el selector número 7 en la posición "BRIDGE MONO ON". Use las entradas del canal 1 y conecte la bocina como se muestra en la figura.

PRECAUCIONES DEL MODO PUENTEADO EN MONO

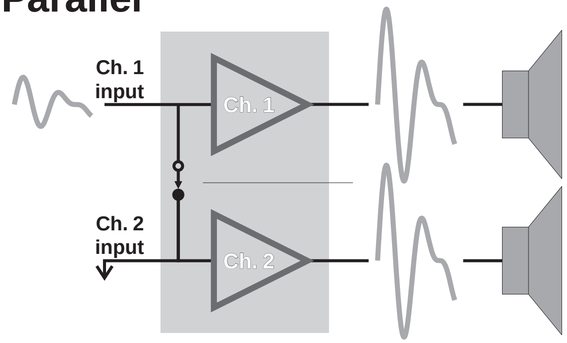
Este modo implica un gran requerimiento para el amplificador y las bocinas. La saturación excesiva puede provocar un silencio para proteger de cualquier daño a la bocina. Asegúrese de que ésta tenga el rango de potencia necesario.

Los voltajes de salida mayores a 100 voltios RMS están disponibles entre las terminales "puenteadas" del amplificador. Los métodos de conexión CLASS 3 (NEC), se especifican de acuerdo a los códigos locales o nacionales, y deben utilizarse para conectar la bocina.

Stereo, bi-amp, 2-channel



Parallel



FEATURES & SETUP

What are the differences among Stereo, Parallel Input, and Bridge Mono modes?

STEREO MODE

This is the "normal" way of using the amplifier, in which each channel is fully independent. Separate signals connect at the inputs, the gain knobs control their respective channels, and separate speakers connect to each output.

Examples:

- Two-channel (stereo) playback.
- Two independent mono signals, such as *main* and *monitor* mixes.
- Bi-amped operation, with the low frequencies in Channel 1 and the highs in Channel 2.

PARALLEL INPUT MODE

This mode is just like Stereo mode, except that the inputs for Channel 1 and Channel 2 are internally connected together. A signal into any input jack will therefore drive both channels directly. Each channel's gain control still functions as usual, and each channel feeds its own speaker load.

You can patch the input signal on to additional amplifiers by using any of the remaining input jacks.

Example:

- One mono signal driving both channels, with independent gain control for each speaker system.

CARACTÉRISTIQUES ET LEUR UTILISATION

Modes stéréo, parallèle et ponté, quelles sont les différences?

MODE STÉRÉO

C'est la façon "normale" d'utiliser l'amplificateur, où les des canaux sont distincts. Des signaux différents peuvent se trouver aux deux entrées, des ajustements de gain, de filtre et de limiteur sont possibles, et des haut-parleurs distincts sont branchés aux deux sorties.

Exemples:

- écoute deux canaux (stéréo)
- deux signaux mono indépendants, tel mix principal et moniteurs de scène
- utilisation en mode bi-amplification, avec les basses fréquences amplifiées par le canal 1 et les hautes par le canal 2 (recquiert l'utilisation d'un séparateur de fréquences électronique vendu séparément)

MODE PARALLÈLE

Similaire au mode stéréo pour les gains, filtres, limiteurs et connexion des haut-parleurs, mais les entrées sont reliées à l'intérieur. Un signal branché à l'une ou l'autre des entrées sera donc amplifié par les deux canaux.

Vous pouvez envoyer le signal vers un autre amplificateur en vous servant de la seconde entrée comme sortie.

Exemple:

- Deux haut-parleurs de même registre dans un système multi-voies

AUSSTATTUNG & EINSTELLUNGEN

Unterschiede zwischen Stereo-, Parallel- und Monobrückenbetrieb

STEREOBETRIEB

Dies ist die "normale" Betriebsart eines Verstärkers, bei der beide Kanäle vollständig unabhängig arbeiten. An den Eingängen liegen separate Signale an, und an den Ausgängen sind unabhängige Lautsprecher angeschlossen.

Beispiele:

- Zweikanal- (Stereo-) Wiedergabe
- Zwei unabhängige Monosignale wie z.B. Summen- und Monitormix
- Biamp-Betrieb, mit den tiefen Frequenzanteilen an Kanal 1 und den Höhen an Kanal 2

PARALLELBETRIEB

Diese Betriebsart entspricht dem Stereobetrieb, außer daß die Eingänge der beiden Kanäle intern zusammengeschaltet werden. Ein Signal, welches an einem der beiden Eingänge anliegt, wird daher beiden Kanälen zugeführt werden. Die Funktion der Regler der beiden Kanäle bleibt vollständig erhalten, und jeder Kanal speist seine eigene Last. Mit Hilfe der freien Eingangsbuchse kann das Eingangssignal auch an weitere Verstärker geleitet werden.

Beispiel:

* Ein Monosignal speist beide Kanäle, mit unabhängiger Verstärkungseinstellung für jedes Lautsprechersystem

CARACTERÍSTICAS Y AJUSTES

¿Cuáles son las diferencias entre los modos Estéreo, Entradas Paralelas y Puenteado en Mono?

MODO ESTÉREO

Esta es la manera "normal" de usar el amplificador, donde cada canal funciona independientemente. Con señales separadas conectadas a las entradas, botones de ganancia que controlan su respectivo canal y bocinas separadas conectadas a cada salida.

Ejemplos:

- Reproducción de dos canales (Estéreo).
- Dos señales mono independientes, como la de la mezcla principal (main) y la de los monitores.
- Operación bi-amplificada, con las frecuencias graves por el canal 1 y las frecuencias agudas por el canal 2.

MODO DE ENTRADAS PARALELAS

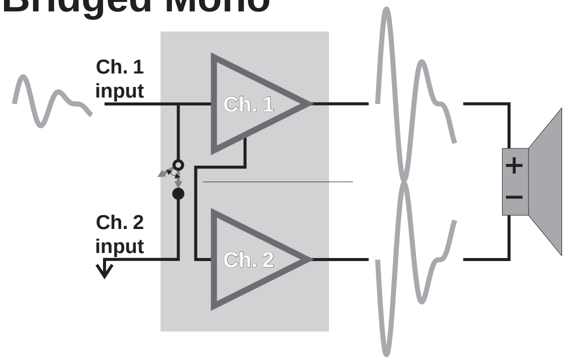
Este modo es como el modo estéreo, excepto que las entradas para el canal 1 y el canal 2, están unidas internamente. Una señal conectada a cualquiera de las entradas alimentará a ambos canales directamente. Los controles de ganancia de cada canal funcionarán de una manera normal, y cada canal lleva su propia carga para la bocina.

Usted puede conectar la señal de entrada a otros amplificadores, usando el conector de entrada que queda libre.

Ejemplo:

- Una señal monofónica que alimenta a ambos canales, con controles de ganacia independientes para cada sistema de bocinas.

Bridged Mono



FEATURES & SETUP

BRIDGE MONO MODE

This mode combines the full power capabilities of both channels into a single speaker system. The amplifier internally re-configures so that both channels operate as a unit. This delivers double the output voltage, resulting in four times the peak power and three times the sustained power into a single 8- or 4-ohm speaker load. The Bridge Mono mode section on page 11 describes the special speaker connection used.

Examples:

- Driving a single 8-ohm speaker with the combined 4-ohm power of both channels.
- Driving a single 4-ohm speaker with the combined 2-ohm power of both channels.

Precautions:

- Bridge Mono mode makes it possible to drive thousands of watts into a single speaker. AC current consumption will usually be higher. Avoid excessive signal level, and make sure the wiring and speaker can handle the power.
- If the load is less than 4 ohms, or prolonged overloads occur, the amplifier will probably mute for several seconds during peaks.
- Do not use 2-ohm loads.

SEE THE ADDITIONAL BRIDGE MONO MODE WARNINGS ON PAGE 11.

CARACTÉRISTIQUES ET LEUR UTILISATION

MODE PONTÉ MONO

Ce mode combine la puissance des deux canaux vers une seule sortie. L'amplificateur est rebranché de façon à ce que les deux canaux fonctionnent en tant qu'unité. Ce branchement double la tension de sortie, ce qui multiplie par un facteur de 4 la puissance en pointe et triple la puissance en continu par rapport à un canal simple dans un charge de 4 ou de 8 ohms. La section Mode ponté mono en page 11 décrit les branchements spéciaux des haut-parleurs en ce mode.

Exemples:

- faire fonctionner un haut-parleur de 8 ohms avec la puissance combinée à 4 ohms des deux canaux
- faire fonctionner un haut-parleur de 4 ohms avec la puissance combinée à 2 ohms des deux canaux

Précautions:

- Le mode ponté mono permet d'envoyer des milliers de watts de puissance vers un seul haut-parleur. La consommation de courant sera plus élevée. S'assurer que le câblage et le haut-parleur peuvent supporter la puissance générée.
- Si la charge (haut-parleur) est de moins de 4 ohms ou que des surtensions prolongées se produisent, il est possible que l'amplificateur passe en mode protection pour quelques secondes lors de surtensions.
- Ne pas utiliser de charges de 2 ohms.

VOIR LES INSTRUCTIONS ADDITIONNELLES POUR UTILISATION EN MODE PONTÉ MONO EN PAGE 11.

AUSSTATTUNG & EINSTELLUNGEN

MONOBRÜCKENBETRIEB

Diese Betriebsart addiert die volle Verstärkung beider Kanäle für eine einzelnes Lautsprechersystem. Der Verstärker wird intern so umkonfiguriert, so daß beide Kanäle wie ein einzelner Kanal funktionieren. Hierdurch steht ungefähr die doppelte Ausgangsspannung, die vierfache Spitzenleistung und ungefähr die dreifache Dauerleistung eines Einzelkanals in eine 8 oder 4Ω Last zur Verfügung. Weitere Erläuterungen zur besonderen Art des Anschlusses der Lautsprecher finden Sie auf Seite 11.

Beispiele:

- Betrieb eines einzelnen 8Ω Lautsprecher mit der addierten 4Ω Leistung beider Kanäle
- Betrieb eines einzelnen 4Ω Lautsprecher mit der addierten 2Ω Leistung beider Kanäle

Vorsichtsmaßnahmen:

- Im Monobrückenbetrieb können tausende Watt Leistung an einen einzelnen Lautsprecher abgegeben werden. Die Stromaufnahme des Verstärkers wird höher als normal sein. Vermeiden Sie daher überhöhte Signalpegel und stellen Sie sicher, daß die Verkabelung und der Lautsprecher die hohe Leistung verarbeiten können.
- Bei Lasten kleiner als 4Ω, oder zu langen Übersteuerungen kann der Verstärker für einige Sekunden abschalten.
- 2Ω Lasten dürfen nicht betrieben werden.

BITTE BEACHTEN SIE DIE WARNHINWEISE AUF SEITE 11.

CARACTERÍSTICAS Y AJUSTES

MODO PUENTEADO EN MONO

Este modo combina la capacidad total de potencia de ambos canales, en un solo sistema de bocinas. El amplificador se reconfigura internamente de manera que ambos canales operen como uno. Esto produce el doble de voltaje de salida, cuatro veces la potencia en picos y tres veces la potencia de sostenimiento en cargas de 8 y 4 ohmios. La sección del modo "Puenteado en Mono" en la página 11 describe la conexión usada para las bocinas.

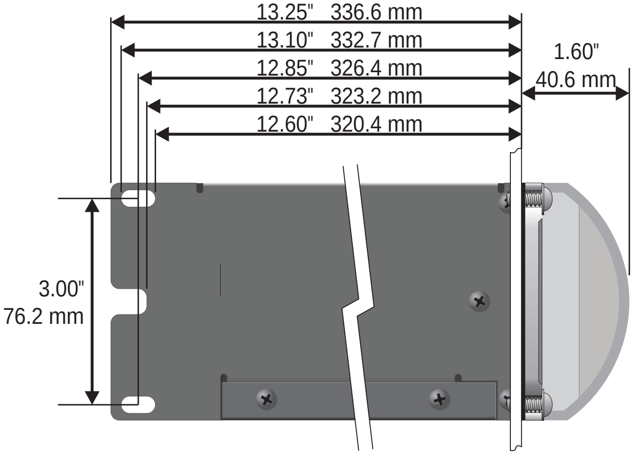
Ejemplos:

- Para alimentar una bocina de 8-ohmios, con la potencia de dos canales de 4-ohms.
- Para alimentar una bocina de 4-ohmios, con la potencia de dos canales de 2-ohms.

Precauciones:

- El modo de puenteado en mono puede mandar miles de watts a una sola bocina. El consumo de corriente alterna (AC) también será mayor. Evite niveles de señales excesivos, y asegurese que el cableado y las bocinas resistan la potencia.
- Si la carga es menor a 4 ohms, ó la saturación es muy frecuente, el amplificador puede enmudecerse (función Mute) durante los picos por varios segundos.
- No use cargas de 2 ohmios.

LEA LAS PRECAUCIONES ADICIONALES PARA EL MODO DE PUENTEADO EN MONO EN LA PÁGINA 11.


13.25" 336.6 mm
13.10" 332.7 mm
12.85" 326.4 mm
12.73" 323.2 mm
12.60" 320.4 mm
1.60"
40.6 mm
3.00"
76.2 mm

INSTALLATION

Use four screws and washers when mounting the amplifier to the front rack rails.

Support the amp at the rear also, especially in mobile and touring use; rear rack mounting ear kits are available from QSC's technical services department or by special order from your dealer or distributor.

INSTALLATION

Utiliser les quatre vis et rondelles de montage pour fixer le devant de l'amplificateur sur les rails de montage.

Il est recommandé de supporter l'arrière de l'amplificateur, tout particulièrement pour applications mobiles et de tournée; des ensembles de support sont disponibles du service à la clientèle de QSC, ou en commande spéciale chez votre marchand ou distributeur local.

EINBAU

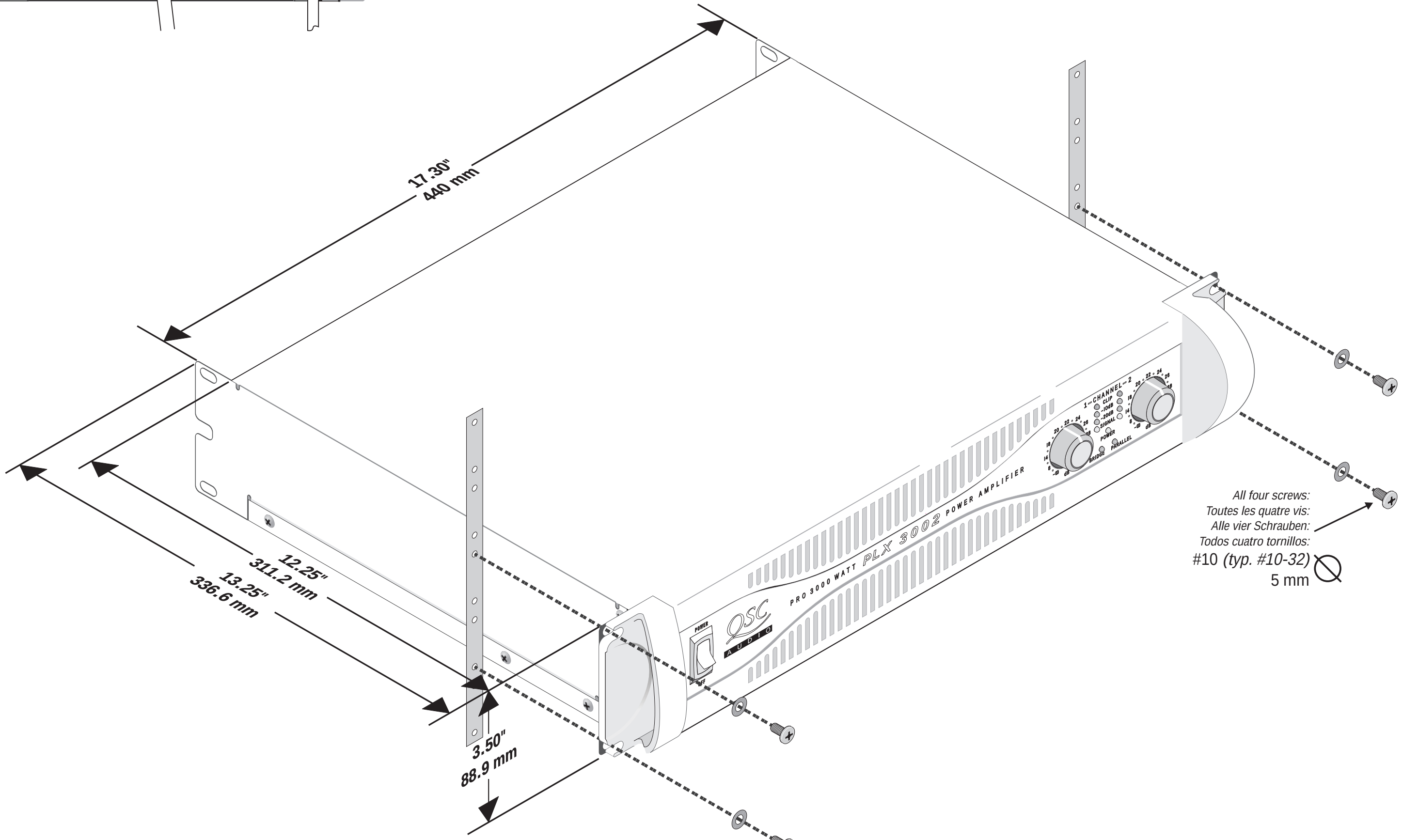
Benutzen Sie vier Befestigungsschrauben und Unterlegscheiben zum Fronteinbau in das Rack.

Stützen Sie den Verstärker auch an seiner Rückseite ab. Dies gilt besonders für den mobilen Einsatz. Rückwärtige Einbausätze können direkt bei QSC oder den jeweiligen Händlern oder Distributoren bestellt werden.

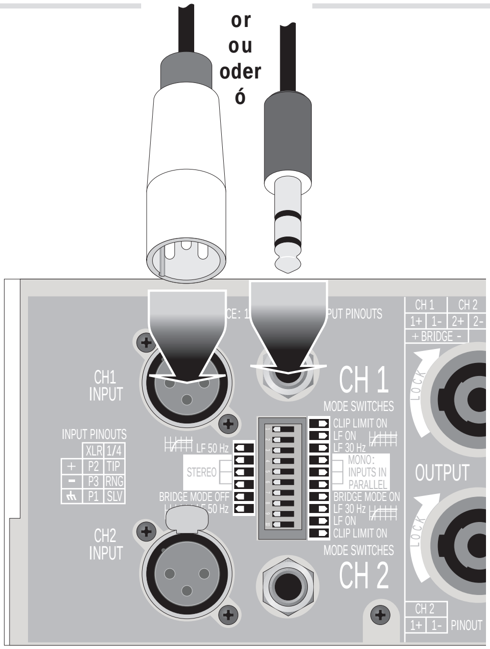
INSTALACIÓN

Utilice cuatro tornillos con rondanas cuando coloque el amplificador en la parte frontal del rack.

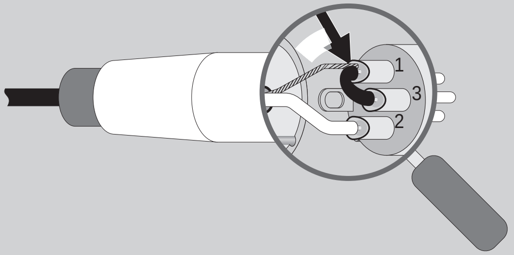
De la misma manera apoye la parte posterior, especialmente cuando vaya a transportar el equipo. Las asas de montaje están disponibles en el departamento de servicios técnicos de QSC o pídalas directamente a su distribuidor.

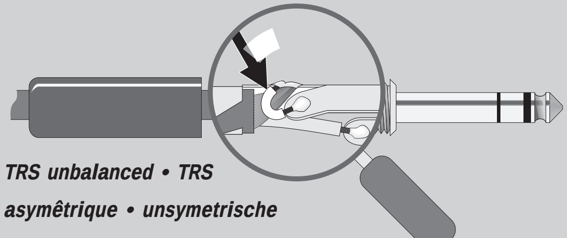

17.30"
440 mm
12.25"
311.2 mm
13.25"
336.6 mm
3.50"
88.9 mm
PRO 3000 WATT PLX 3002 POWER AMPLIFIER
QSC AUDIO
POWER

All four screws:
Toutes les quatre vis:
Alle vier Schrauben:
Todos cuatro tornillos:
#10 *(typ. #10-32)*
5 mm



XLR unbalanced • XLR asymêtrique • unsymetrische XLR • XLR no balanceado





TRS unbalanced • TRS asymêtrique • unsymetrische Stereoklinke • TRS no balanceado

CONNECTIONS

Inputs

Each channel has active balanced XLR and ¼-inch (6.3 mm) inputs wired in parallel. The input impedance is 12 KΩ balanced, 6 KΩ unbalanced.

Balanced signals are less prone to AC hum, but unbalanced signals can be suitable for short cable runs. The signal source's output impedance should be less than 600Ω to avoid high frequency loss in long cables.

Balanced inputs: Use the XLR or ¼-inch (6.3 mm) TRS input jacks.

Unbalanced inputs: Connect the unused side of the balanced input to ground, as shown below left. A tip-sleeve ¼-inch (6.3 mm) connector will correctly terminate the unused side of the input without modification.

For two-channel (stereo) operation, use the inputs for both Channel 1 and Channel 2; for parallel or bridged mono operation, use the Channel 1 input. See the section on operating modes for more explanation. To patch the audio signal to other amps (parallel and bridged modes only), see the instructions for using parallel inputs on page 9.

CONNEXIONS

Entrées

Chaque canal est équipé d'une entrée symétrique sur prises XLR et ¼" (6.3 mm) branchées en parallèle. L'impédance d'entrée est de 12 KΩ symétrique, 6 KΩ asymétrique.

Les signaux reçus en mode symétrique sont moins sujets aux ronflements dû aux interférences de réseau électrique (hum), alors que les branchements asymétriques peuvent convenir pour les courtes distances. L'impédance de sortie de la source de signal devrait être de moins de 600Ω afin de réduire les pertes en hautes fréquences avec de longs câbles.

Entrées symétriques: Utilisez la prise XLR ou ¼" (6.3 mm).

Entrées asymétriques: Connectez le côté non-utilisé de l'entrée symétrique au châssis, tel qu'illustré ci-contre. Une fiche ¼" (6.3 mm) mono (Tip Sleeve) fermera le circuit sans modification.

En utilisation deux canaux (stéréo), utilisez les prises des canaux 1 et 2; en mode parallèle ou ponté mono, utilisez la prise du canal 1. Voir les explications dans les sections appropriées du présent manuel. Pour relier le signal audio de cet amplificateur vers d'autres amplificateurs (mode parallèle et ponté mono seulement), voyez les instructions en page 9.

ANSCHLÜSSE

Eingänge

Jeder Kanal verfügt über symmetrische XLR- und 6,3 mm Stereoklinkeneingänge. Die Eingangsimpedanz beträgt symmetrisch 12 KΩ, unsymmetrisch 6 KΩ.

Symmetrische Signale verursachen weniger Brummproblem, wobei unsymmetrische Verbindungen allerdings bei kurzen Kabelwegen häufig ausreichen. Die Quellimpedanz sollte dabei weniger als 600Ω betragen, um einen Höhenverlust bei längeren Kabelwegen zu vermeiden.

Symmetrische Eingänge: Verwenden Sie die XLR oder 6,3mm Stereoklinkenbuchsen

Unsymmetrische Eingänge: Verbinden Sie den ungenutzten Pin des symmetrischen Steckers mit Masse (siehe Abbildung). Bei einem Mono-Klinkenstecker ist keine Änderung nötig.

Im 2-Kanalbetrieb (Stereo), verwenden Sie bitte beide Eingänge; im Parallel-, bzw. Monobrückenbetrieb verwenden Sie nur Eingang 1. Bitte beachten sie die jeweiligen Beschreibungen der verschiedenen Betriebszustände. Um das Signal an weitere Verstärker zu leiten (nur im Parallel-, bzw. Monobrückenbetrieb), beachten Sie bitte die Beschreibungen auf Seite 9.

CONEXIONES

Entradas

Cada canal tiene entradas activas balanceadas con conectores XLR y ¼" (6.3 mm), cableadas en paralelo. La impedancia de entrada es de 12 KΩ en la balanceada, 6 KΩ en la no balanceada.

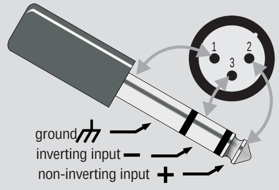
Las señales balanceadas son menos propensas a inducir zumbidos debido a la corriente alterna, pero para cables cortos se pueden usar señales no balanceadas. La impedancia de salida debe ser menor de 600 ohms para prevenir la pérdida de frecuencias agudas en cables largos.

Entradas balanceadas: Use conectores XLR o TRS de ¼" (6.3 mm).

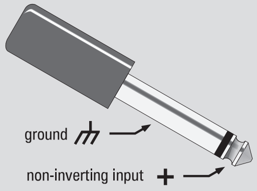
Entradas no balanceadas: Conecte el lado libre de la salida balanceada a tierra, como se muestra a la izquierda. Con un conector tipo tip-sleeve de ¼" (6.3mm) se llevará a cabo la terminación apropiada del conductor sin usar de la entrada sin ninguna modificación.

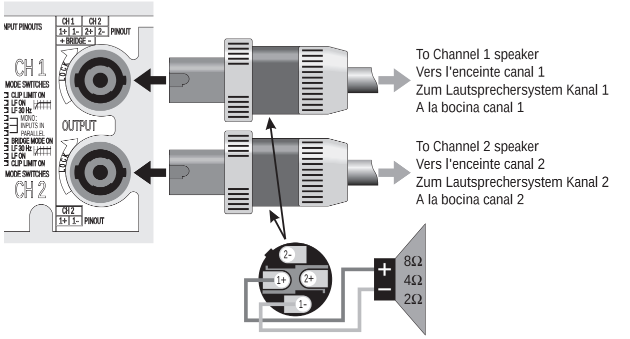
Para operación en dos canales (estéreo), use las entradas de los canales 1 y 2. Para operación en paralelo o puenteada en mono, use la entrada del canal 1. Lea la sección de los modos de operación para una explicación más detallada. Para enviar la señal de audio a otros amplificadores (sólo en los modos paralelo o puenteado en mono), lea las instrucciones para entradas paralelas en la página 9.

- Balanced
- Symétrique
- Symmetrisch
- Balanceado



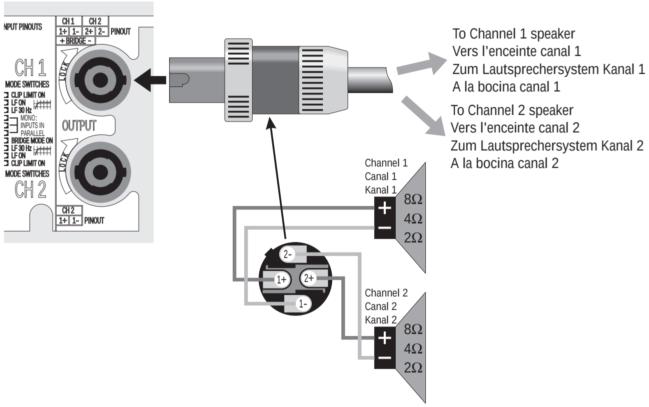
- Unbalanced
- Asymétrique
- Unsymmetrisch
- No balanceado




INPUT PINOUTS
CH 1
CH 2
+ BRIDGE –
PINOUT
MODE SWITCHES
OUTPUT
To Channel 1 speaker
Vers l'enceinte canal 1
Zum Lautsprechersystem Kanal 1
A la bocina canal 1
To Channel 2 speaker
Vers l'enceinte canal 2
Zum Lautsprechersystem Kanal 2
A la bocina canal 2
2-
1+
2+
1-
+
–
8Ω
4Ω
2Ω

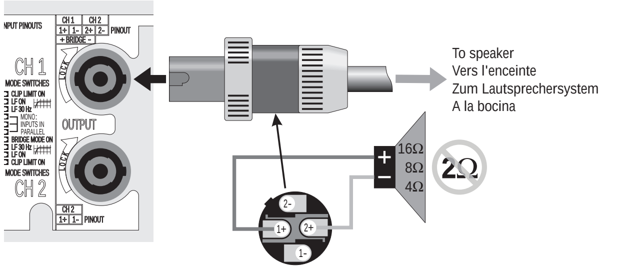
2 channels/canals/Kanäle/canals & 2 Speakons

(Stereo, bi-amp, or parallel mode; Modes stéreo, bi-amp ou parallèle; Stereo-, Bi-amp- oder Parallelbetrieb; Modos estéreo, bi-amp o paralelo)


INPUT PINOUTS
CH 1
CH 2
+ BRIDGE –
PINOUT
MODE SWITCHES
OUTPUT
To Channel 1 speaker
Vers l'enceinte canal 1
Zum Lautsprechersystem Kanal 1
A la bocina canal 1
To Channel 2 speaker
Vers l'enceinte canal 2
Zum Lautsprechersystem Kanal 2
A la bocina canal 2
Channel 1
Canal 1
Kanal 1
+
–
8Ω
4Ω
2Ω
2-
1+
2+
1-
Channel 2
Canal 2
Kanal 2
+
–
8Ω
4Ω
2Ω

2 channels/canals/Kanäle/canals & 1 Speakon

(Stereo, bi-amp, or parallel mode; Modes stéreo, bi-amp ou parallèle; Stereo-, Bi-Amp- oder Parallelbetrieb; Modos estéreo, bi-amp o paralelo)


INPUT PINOUTS
CH 1
CH 2
+ BRIDGE –
PINOUT
MODE SWITCHES
OUTPUT
To speaker
Vers l'enceinte
Zum Lautsprechersystem
A la bocina
+
–
16Ω
8Ω
4Ω
2Ω
2-
1+
2+
1-

Bridged mono • Mono ponté • Monobrückenbetrieb • Mono puente

CONNECTIONS

Speakon™ Outputs

The PLX amplifier offers a choice of output connections, with two Neutrik NL4MD Speakon jacks and two pairs of "touchproof" binding posts.

The Speakon connector is designed specially for high-power speaker connections. It locks in place, prevents shock hazard, and assures the correct polarity.

The upper Speakon jack has both Channel 1 and Channel 2 outputs, so it is especially useful for parallel, bi-amp, or bridged mono operation (see bridged mono operating precautions on page 12). The other Speakon carries only Channel 2's output. See the illustrations at left.

For easier insertion, use the newer-style NL4FC Speakon connectors with quicklock thumb latches.

SPEAKER CABLING

Larger wire sizes and shorter lengths minimize both loss of power and degradation of damping factor. Do not place speaker cables next to input wiring.

WARNING: To prevent electric shock, do not operate the amplifier with any of the conductor portion of the speaker wire exposed.

CONNEXIONS

Sorties Speakon™

Cet amplificateur PLX offre un choix de connecteurs de sortie, avec deux prises Neutrik Speakon NL4MD et deux paires de bornes à écrou protégées.

Les prises Speakon sont conçues spécifiquement pour la connexion de haut-parleurs de haute puissance. Elles verrouillent en place, préviennent les risques de choc électrique, et assurent le branchement avec polarité correcte.

La prise Speakon du haut porte les signaux de sortie des deux canaux, elle est donc très utile lors de branchement en modes parallèle, bi-amplifié, ou ponté mono (voir les instructions et précautions à suivre pour le branchement en mode ponté mono en page 12). La prise du bas ne porte que le signal du canal 2. Voir les illustrations ci-contre.

Pour insertion plus facile, utiliser la nouvelle génération des Speakon NL4FC avec loquet à ressort.

CÂBLAGE DU HAUT-PARLEUR

Les câbles de fort calibres et les courtes distances minimisent à la fois les pertes de puissance et la dégradation du coefficient d'amortissement. Evitez de placer les câbles de haut-parleurs à proximité des câbles d'entrée.

AVERTISSEMENT: Afin de prévenir les risques de choc électrique, ne pas utiliser l'amplificateur si une portion de conducteur du fil de haut-parleur est exposée.

ANSCHLÜSSE

Speakon™ Ausgänge

Der PLX Verstärker stellt mehrere Ausgangsanschlüsse zur Verfügung: zwei Neutrik NL4MD Speakonbuchsen und zwei Paar berührungssichere Schraubklemmen.

Der Speakonanschluß wurde speziell für den Anschluß von Hochleistungslautsprechern konzipiert. Er rastet ein, verhindert elektrischen Schlag und sorgt für die richtige Polarität.

Die obere Speakonbuchse enthält beide Kanäle, so daß sie besonders für Parallel-, Biamp- oder Monobrückenbetrieb geeignet ist (bitte beachten Sie die Vorsichtsmaßnahmen wie auf Seite 12 beschrieben). Die andere Speakonbuchse enthält nur die Signale von Kanal 2 (siehe auch nebenstehende Zeichnung).

Für eine leichtere Anbindung, verwenden Sie die NL4FC-Speakonstecker der neueren Generation.

LAUTSPRECHERKABEL

Dickere und kürzere Kabel verringern sowohl Leistungsverlust wie eine Verschlechterung des Dämpfungsfaktors. Verlegen Sie keine Ausgangskabel neben den Eingangskabeln.

WARNUNG: Um elektrische Schläge zu vermeiden, sollte der Verstärker nicht betrieben werden, wenn blanke Kabelenden sichtbar sind.

CONEXIONES

Salidas Speakon™

El amplificador PLX ofrece varias opciones de salida, con dos conectores Neutrik Speakon NL4MD y dos pares de terminales con tornillo.

El conector Speakon está diseñado especialmente para conexiones de bocinas de alta potencia. Se ajusta fijamente, previene cortos eléctricos y asegura una correcta polaridad.

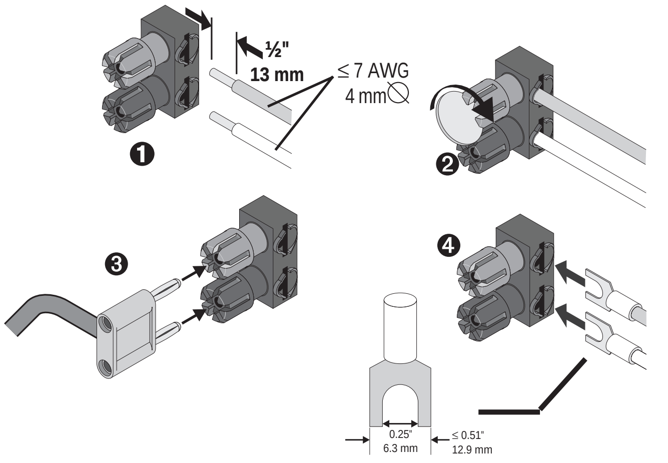
El conector Speakon superior tiene las salidas de los canales 1 y 2, y se usa para una operación en paralelo, biamplificada, o puenteada en mono (lea las recomendaciones de la página 12). El otro conector Speakon sólo tiene la salida del canal 2. Observe las ilustraciones a la izquierda.

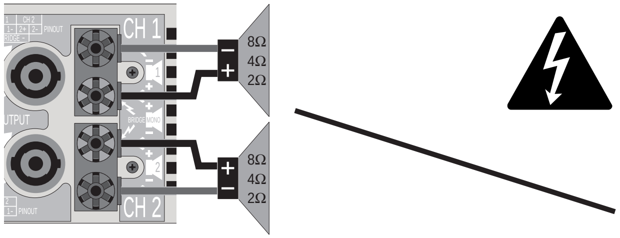
Para facilitar la inserción de los conectores, le recomendamos que use los nuevos conectores Speakon NL4FC con retén.

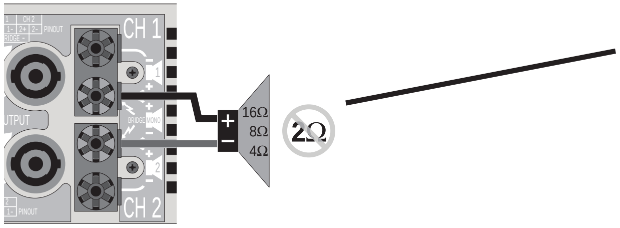
CABLEADO DE BOCINAS

Los cables gruesos y los trayectos cortos minimizan la pérdida de potencia y degradación por el factor de amortiguación. No coloque los cables de las bocinas cercanos a los de entrada.

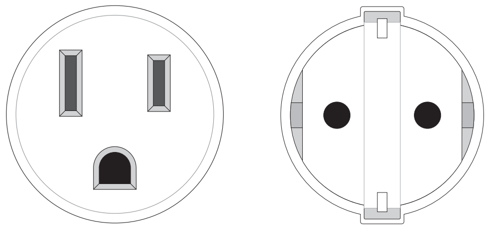
¡AVISO! Para evitar una descarga eléctrica, no opere el amplificador si alguno de los cables de la bocina está expuesto.







Outlets • Prises de courant • Netzanschluß • Tomas de corriente



120V

220–240V

CONNECTIONS

Binding post outputs

❶ Strip back insulation not more than 13 mm (½ inch).

❷ Insert wire fully so that none of the conductor is exposed; tighten barrel (use coin if necessary).

❸ Non-European models only.

❹ Spade lugs must have insulated barrels to prevent electric shock.

WARNING: To prevent electric shock, do not operate the amplifier with any of the conductor portion of the speaker wire exposed.

Connections for stereo and parallel operations

Connections for bridged mono operation. See bridged mono operating precautions on page 11.

SPEAKER CABLING

Larger wire sizes and shorter lengths minimize both loss of power and degradation of damping factor. Do not place speaker cables next to input wiring.

Operating voltage (AC mains)

Make sure you connect the amplifier to the correct AC line voltage, which is shown on the serial number label. Connecting to the wrong line voltage is dangerous and may damage the amplifier.

CONNEXIONS

Bornes à écrou

❶ Dénuder le fil sur un maximum de 13 mm (½ puce).

❷ Insérer le fil de façon à ce qu'aucune partie du conducteur ne soit exposée; serrer le corps du connecteur (utiliser une pièce de monnaie si nécessaire).

❸ Modèles non-européens seulement.

❹ Les cosses à fourche devraient avoir un corps isolé afin de prévenir les risques de choc électrique.

AVERTISSEMENT: Afin de prévenir les risques de choc électrique, ne pas utiliser l'amplificateur si une portion de conducteur du fil de haut-parleur est exposée.

Connexions pour utilisation en modes stéréo et parallèle.

Connexions pour utilisation en mode ponté mono. Voir les instructions et précautions à suivre pour le branchement en mode ponté mono en page 12.

CÂBLAGE DU HAUT-PARLEUR

Les câbles de fort calibres et les courtes distances minimisent à la fois les pertes de puissance et la dégradation du coefficient d'amortissement. Evitez de placer les câbles de haut-parleurs à proximité des câbles d'entrée.

Tension d'utilisation (alimentation CA)

Vous devez vous assurer de raccorder l'amplificateur à une source de courant alternatif de tension correcte, tel qu'indiqué sur l'étiquette du numéro de série. Connecter à une source de courant autre qu'indiqué est dangereux et pourrait endommager l'amplificateur.

ANSCHLÜSSE

Anschlussklemmen

❶ Das Kabelende nicht länger als 13 mm abisolieren.

❷ Das abisolierte Kabelende komplett einführen, ohne daß blanker Draht sichtbar bleibt; Klemme festdrehen. Münze benutzen falls notwendig.

❸ Nur bei außereuropäischen Gerätetypen.

❹ Um die Gefahr elektrischer Schläge zu vermeiden, müssen Kabelklemmschuhe isolierte Klemmen aufweisen.

WARNUNG: Um elektrische Schläge zu vermeiden, sollte der Verstärker nicht betrieben werden, wenn blanke Kabelenden sichtbar sind.

Anschlüsse für Stereo-, bzw. Parallelbetrieb.

Anschluß im Monobrückenbetrieb. Bitte beachten Sie die Erläuterungen auf Seite 11.

LAUTSPRECHERKABEL

Dickere und kürzere Kabel verringern sowohl Leistungsverlust wie eine Verschlechterung des Dämpfungsfaktors. Verlegen Sie keine Ausgangskabel neben den Eingangskabeln.

Netzanschluß

Schließen Sie den Verstärker nur an die richtige Netzspannung an, die auf dem Etikett mit der Seriennummer aufgedruckt ist. Ein Anschluß an die falsche Netzspannung kann zu Beschädigungen am Verstärker führen.

CONEXIONES

Terminales con tornillo

❶ Remueva el aislamiento no más de 13 mm (½ pulgado).

❷ Inserte el cable hasta que el conductor se pueda ver. Asegure la rosca con fuerza (si es necesario use una moneda).

❸ Unicamente para modelos no-europeos.

❹ Los conectores tipo herradura deben estar aislados apropiadamente para prevenir un corto eléctrico.

¡AVISO! Para evitar una descarga eléctrica, no opere el amplificador si alguno de los cables de la bocina está expuesto.

Conexiones para operación en estéreo o paralelo.

Conexiones para operación de puenteada en mono. Lea las recomendaciones de la página 11.

CABLEADO DE BOCINAS

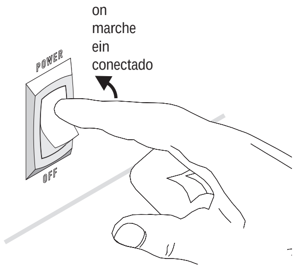
Los cables gruesos y los trayectos cortos minimizan la pérdida de potencia y degradación por el factor de amortiguación. No coloque los cables de las bocinas cercanos a los de entrada.

Voltaje de operación (alimentación CA)

Asegúrese de conectar el amplificador a una línea de corriente alterna (AC) con el voltaje adecuado, el cual aparece en la etiqueta con el número de serie. Conectarlo a una línea de voltaje diferente es peligroso y puede dañarlo.

OPERATION

AC power switch

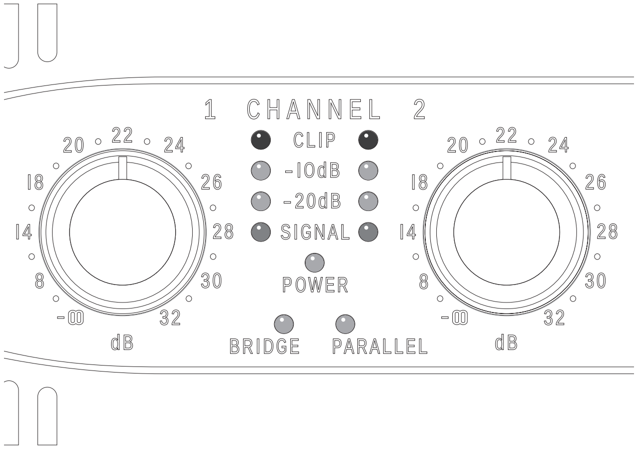


Before applying power, check all connections and turn down the gain controls. The "soft start" sequence starts with the ***POWER*** indicator LED at half brightness. A couple seconds later the fan starts and the amplifier cycles through one second of protective muting, indicated by the ***CLIP*** LEDs glowing bright red. The ***POWER*** indicator then changes to full brightness and the amplifier is ready.

Gain controls

The gain controls are detented for repeatable adjustment. The actual voltage gain of the amplifier is shown in dB. Maximum gain is 40×, or +32 dB.

LED indicators



At full brightness, the green ***POWER*** LED indicates that the amplifier is operating. Half brightness means the amp is in its startup sequence.

As the signal increases, the green ***SIGNAL, -20dB,*** and ***-10dB*** LED indicators light respectively at 0.1%, 1% and 10% of full power.

The red ***CLIP*** LED indicator flashes during overload (clipping). A bright, steady glow indicates protective muting. If this occurs during use, see Troubleshooting.

The yellow ***BRIDGE*** LED indicates the amp is in bridged mono mode.

The yellow ***PARALLEL*** LED indicates the Parallel Input switches are set.

UTILISATION

Interrupteur d'alimentation CA

Avant de mettre l'amplificateur en marche, vérifiez toutes les connexions et fermez les contrôles de gain. A la mise en marche, la fonction "Soft start" est activée, avec comme indication l'allumage de la DEL ***POWER*** à demi-intensité. Quelques secondes plus tard le ventilateur démarre et l'amplificateur passe en mode protection pour une seconde, tel qu'indiqué par la DEL ***CLIP*** qui passe au rouge. La DEL ***POWER*** passe alors à pleine intensité et l'amplificateur est prêt à fonctionner.

Contrôles de gain

Les contrôles de gain à taquets permet les réajustements répétés. Le gain en tension de l'amplification est indiqué en dB. Le gain maximum est de 40×, soit +32 dB.

Indicateurs DEL

A pleine intensité, la DEL (diode électroluminescente) verte de ***POWER*** indique que l'amplificateur est en marche. A demi-intensité, elle indique que l'amplificateur est en mise en marche.

Au fur et à mesure que le niveau de signal augmente, les DEL vertes ***SIGNAL***, ***-20dB***, et ***-10dB*** allument respectivement à 0.1%, 1%, et 10% de la pleine puissance de l'amplificateur.

La DEL ***CLIP*** clignote lors de surcharges (écrêtage). La DEL reste allumée à pleine intensité pour indiquer la mise en sourdine par le circuit de protection. Si cette condition se présentait lors de l'utilisation, voir la section dépannage de ce manuel.

La DEL jaune ***BRIDGE*** indique que l'amplificateur a été réglé en mode ponté mono.

La DEL jaune ***PARALLEL*** indique que l'amplificateur a été réglé en mode parallèle.

BETRIEB

Netzschalter

Bevor Sie einschalten, überprüfen Sie alle Verbindungen und drehen Sie die Verstärkung zurück. Die Softstart-Sequenz beginnt mit halber Helligkeit der ***POWER*** LED. Einige Sekunden später beginnt der Ventilator zu laufen und der Verstärker schaltet für etwa eine Sekunde stumm, wobei die rote ***CLIP*** LED hell aufleuchtet. Danach erscheint die ***POWER***-Anzeige in voller Helligkeit und der Verstärker ist betriebsbereit.

Verstärkungsregler

Die Regler wurden als Rastpotentiometer ausgelegt, um reproduzierbare Einstellungen zu erlauben. Die tatsächliche Verstärkung wird in dB angezeigt. Volle Verstärkung entspricht 40×, oder 32 dB.

LED-Anzeige

Bei voller Helligkeit zeigt die grüne ***POWER*** LED die Betriebsbereitschaft des Verstärkers an. Halbe Helligkeit bedeutet, daß sich der Verstärker in der Startsequenz befindet.

Bei steigendem Eingangssignal, leuchten die grünen ***SIGNAL***, ***-20dB*** und ***-10dB*** LEDs nacheinander auf und zeigen 0,1%, 1% und 10% der möglichen vollen Leistung an.

Die rote ***CLIP*** LED leuchtet während Übersteuerungen (Clipping) auf. Ein helles, gleichmäßiges Leuchten zeigt außerdem schützendes Stummschalten an. Falls dieser Zustand während des Betriebs auftritt, lesen Sie bitte den Abschnitt Fehlerbehebung.

Die gelbe ***BRIDGE*** LED leuchtet auf, wenn der Verstärker Monobrücke betrieben wird.

Die gelbe ***PARALLEL*** LED zeigt an, daß die Parallel Input Schalter eingestellt wurden.

OPERACIÓN

Interruptor de encendio

Antes de encender el equipo, revise las conexiones y baje los controles de ganancia. La secuencia de encendido "suave" inicia con el indicador LED ***POWER*** a media luz. Un par de segundos después el ventilador se enciende y el amplificador hace un silencio momentáneo de protección, que se puede visualizar en los indicadores LED rojos de ***CLIP***. Después el indicador ***POWER*** se enciende completamente y el amplificador está listo para operar.

Controles de ganancia

Los controles de ganancia han sido diseñados para un ajuste continuo. El voltaje de ganancia del amplificador aparece en decibeles. La ganancia máxima es de 40×, ó +32 dB.

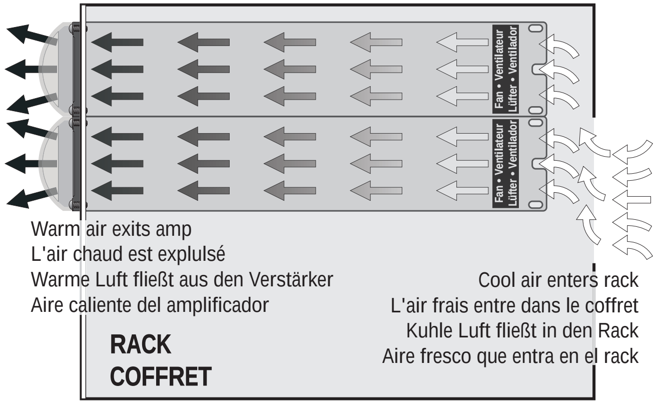
Indicadores LED

La luz verde en el LED de ***POWER*** indica que el amplificador está en uso. Y a media luz, que está en la secuencia de encendido. A medida de que la señal aumenta, los LED indicadores verdes de ***SIGNAL***, los de ***-20 dB*** y ***-10 dB***, se illuminan respectivamente al 0.1%, 1%, y 10% de la potencia máxima.

El indicador LED rojo de ***CLIP***, parpadea cuando hay saturación *(clips)*. Cuando permanece encendido indica "enmudecimiento" (función *Mute*) de protección. Si esto ocurre durante el uso, lea la sección Busqueda de Averías.

El LED amarillo de ***BRIDGE*** indica que el amplificador está en modo de puenteado en mono.

El LED amarillo de ***PARALLEL*** indica que los interruptores de entrada paralela han sido activados.


Fan • Ventilateur
Lüfter • Ventilador
Fan • Ventilateur
Lüfter • Ventilador
Warm air exits amp
L'air chaud est explulsé
Warme Luft fließt aus den Verstärker
Aire caliente del amplificador
Cool air enters rack
L'air frais entre dans le coffret
Kuhle Luft fließt in den Rack
Aire fresco que entra en el rack
RACK
COFFRET

OPERATION

Fan cooling

The fan speed varies automatically to maintain safe internal temperatures. Keep the front and rear vents clear to allow full air flow.

Hot air exhausts out the front of the amp so it does not heat the interior of the rack. Make sure that plenty of cool air can enter the rack, especially if there are other units which exhaust hot air into it.

Safe operating levels

The amp's protective muting system guards against excessive internal temperatures. With normal ventilation and 4- to 8-ohm loads, the amplifier will handle any signal level including overdrive—but make sure that the speakers can handle the full power! However, lower load impedances and higher signal levels produce more internal heating. Into 2-ohm loads, frequent or prolonged clipping (indicated by constant flashing of the red ***CLIP*** LED) may trigger power cutback or even protective muting.

Bridged mono mode doubles the output impedance of the amp; 4 ohms is the minimum load impedance. Heavy clipping may cause muting. If this happens, see Troubleshooting, page 23.

UTILISATION

Ventilation

Le ventilateur à vitesse variable s'ajuste automatiquement pour maintenir une température interne raisonnable. Veillez à garder les entrées et sorties d'air libres d'obstructions afin de permettre libre circulation d'air.

L'air chaud sort par le devant de l'amplificateur afin de pas surchauffer l'intérieur du coffret d'équipement. Assurez vous que beaucoup d'air frais puisse entrer dans le coffret, tout particulièrement si vous utilisez d'autres appareils qui soufflent leur air chaud à l'intérieur du coffret.

Niveaux d'utilisation sécuritaire

Le circuit de protection de l'amplificateur entre en fonction et met le signal en sourdine lors de température excessive. Avec une charge de 4 ou de 8 ohms et avec une ventilation normale, l'amplificateur fonctionnera sans arrêt avec tout signal, incluant les surcharges-assurez vous cependant que les haut-parleurs peuvent accepter toute la puissance de l'amplificateur. Il est possible que des charges d'impédance plus faible ou que des niveaux de signal plus élevés génèrent plus de chaleur. Ainsi, une charge de 2 ohms accompagnée de surcharges constantes, tel qu'indiqué par l'allumage continuel de la DEL ***CLIP***, pourrait amener l'amplificateur à réduire le niveau de puissance, voire à amener le passage en mode protection, avec mise en sourdine.

Le mode ponté mono double l'impédance de sortie de l'amplificateur; 4Ω devient donc la charge minimum. L'écrêtage continuel pourrait entraîner la mise en sourdine. Si cette condition se présentait lors de l'utilisation, voir la section dépannage de ce manuel.

BETRIEB

Lüfterkühlung

Die Lüftergeschwindigkeit ist geregelt, um immer einen sicheren Betrieb zu gewährleisten. Blockieren Sie niemals die Lufteinlaß-, bzw. -ausströmöffnungen.

Die heiße Kühlluft verläßt tritt aus der Front des Verstärkers aus, damit das Rack nicht aufgeheizt wird. Stellen Sie sicher, daß ausreichend kühle Luft in das Rack gelangen kann. Dies gilt besonders dann, wenn andere Geräte warme Kühlluft in das Rack leiten.

Betriebspegel

Die Schutzabschaltung des Verstärkers schützt vor übermäßigen internen Temperaturen. Bei normaler Lüftung und Lasten von 4–8Ω kann der Verstärker jedes Signal, einschließlich Übersteuerungen problemlos verarbeiten—stellen Sie aber sicher, daß die angeschlossenen Lautsprecher die Leistung verarbeiten können.

Niedrigere Impedanzen und höhere Signalpegel führen zu größerer Abwärme. Häufige oder lange Übersteuerungen (erkennbar durch dauerndes Aufflackern der roten ***CLIP*** LED) mit Lasten von 2Ω können eine Leistungsverringerung oder sogar eine Abschaltung der Endstufe zur Folge haben.

Der Monobrückenbetrieb verdoppelt die Ausgangsimpedanz des Verstärkers. Daher dürfen in dieser Betriebsart keine Lasten kleiner 4Ω angeschlossen werden. Erhebliche Übersteuerungen könne ein Abschalten zur Folge haben. In diesem Fall lesen Sie bitte den Abschnitt Fehlerbehebung (Seite 23).

OPERACIÓN

Ventilación

La velocidad del ventilador varía automáticamente para mantener la temperatura interna adecuada. Mantenga las rejillas de ventilación delanteras y posteriores despejadas para que el aire pueda circular.

El aire caliente sale por la parte frontal del amplificador para no calentar el rack. Asegúrese de que circule aire fresco en el rack, especialmente si hay otras unidades que generan aire caliente hacio adentro.

Niveles adecuados de operación

El sistema de "enmudecimiento" (función *Mute*) de protección, se activa con temperaturas altas internas. Con una ventilación normal y cargas de 4 y 8 ohmios, el amplificador puede manejar cualquier nivel de señal incluyendo la saturación—pero verifique que sus bocinas puedan manejar esa potencia. Sin embargo, las bajas impedancias y los altos niveles de señal producen calor interno. En cargas de 2 ohmios, con *'clipeo'* frecuente o prolongado (indicado por el parpadeo del LED ***CLIP***) puede ocurrir un corte de corriente o un "enmudecimiento" de protección.

El modo de puenteado en mono duplica la impedancia de salida del amplificador; 4 ohmios es la mínima impedancia de carga. La saturación extrema puede causar paro total. Si ésto sucede lea la sección Busqueda de Averías, en la página 23.

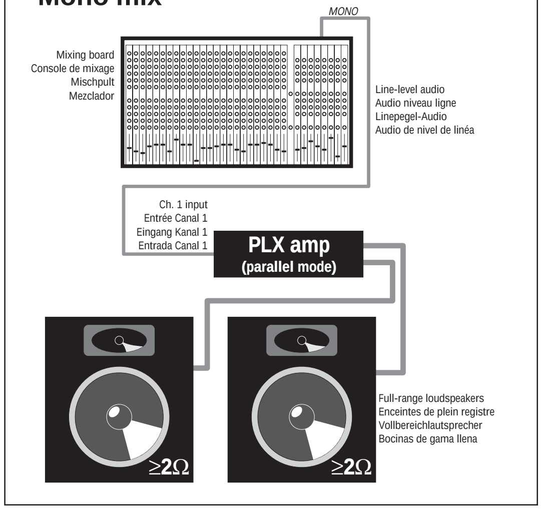
APPLICATIONS
Sound reinforcement

APPLICATIONS
Sonorisation

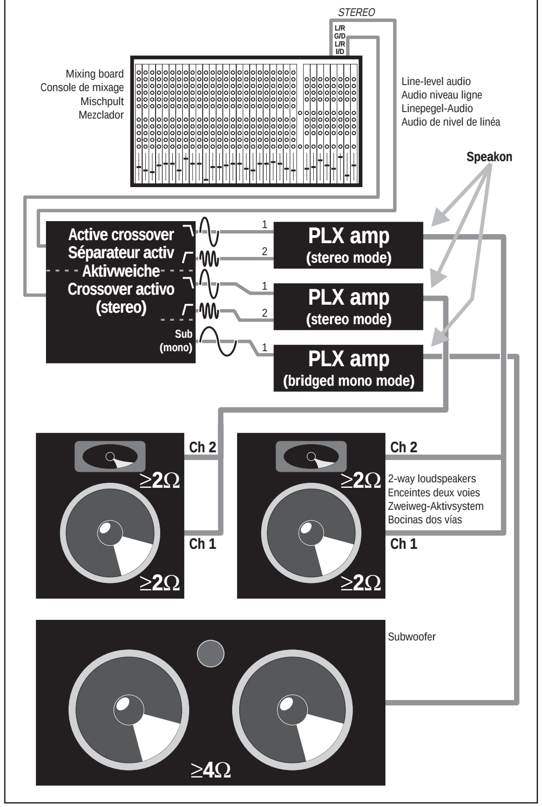
ANWENDUNGS-BEISPIELE
Beschallung

APLICACIONES
Sonido en vivo

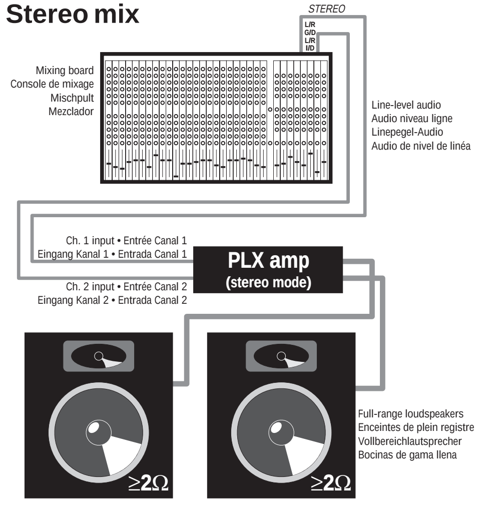
Mono mix


MONO
Mixing board
Console de mixage
Mischpult
Mezclador
Line-level audio
Audio niveau ligne
Linepegel-Audio
Audio de nivel de linéa
Ch. 1 input
Entrée Canal 1
Eingang Kanal 1
Entrada Canal 1
PLX amp
(parallel mode)
Full-range loudspeakers
Enceintes de plein registre
Vollbereichlautsprecher
Bocinas de gama llena
≥2Ω
≥2Ω

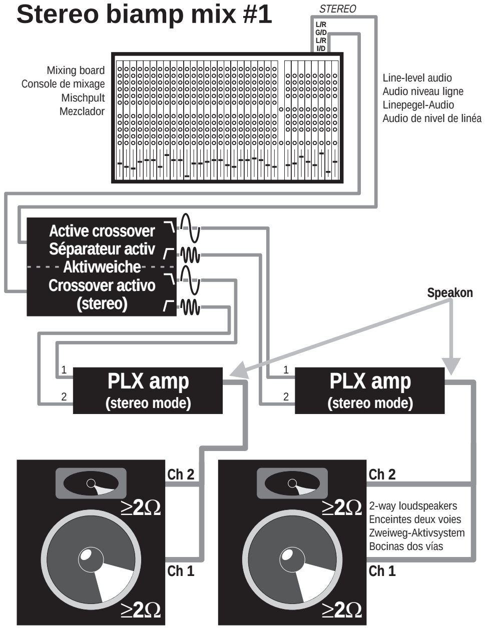
Stereo biamp mix #1, + subwoofer


STEREO
L/R
G/D
L/R
I/D
Mixing board
Console de mixage
Mischpult
Mezclador
Line-level audio
Audio niveau ligne
Linepegel-Audio
Audio de nivel de linéa
Speakon
Active crossover
Séparateur activ
Aktivweiche
Crossover activo
(stereo)
Sub
(mono)
1
2
1
2
1
PLX amp
(stereo mode)
PLX amp
(stereo mode)
PLX amp
(bridged mono mode)
Ch 2
Ch 1
Ch 2
Ch 1
≥2Ω
≥2Ω
≥2Ω
≥2Ω
2-way loudspeakers
Enceintes deux voies
Zweiweg-Aktivsystem
Bocinas dos vías
Subwoofer
≥4Ω

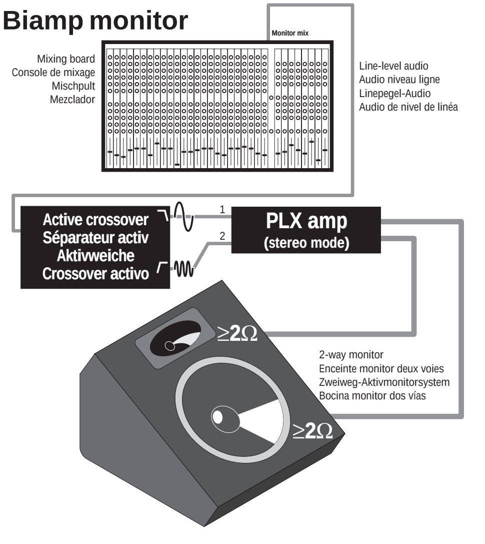
Stereo mix


STEREO
L/R
G/D
L/R
I/D
Mixing board
Console de mixage
Mischpult
Mezclador
Line-level audio
Audio niveau ligne
Linepegel-Audio
Audio de nivel de linéa
Ch. 1 input • Entrée Canal 1
Eingang Kanal 1 • Entrada Canal 1
PLX amp
(stereo mode)
Ch. 2 input • Entrée Canal 2
Eingang Kanal 2 • Entrada Canal 2
Full-range loudspeakers
Enceintes de plein registre
Vollbereichlautsprecher
Bocinas de gama llena
≥2Ω
≥2Ω

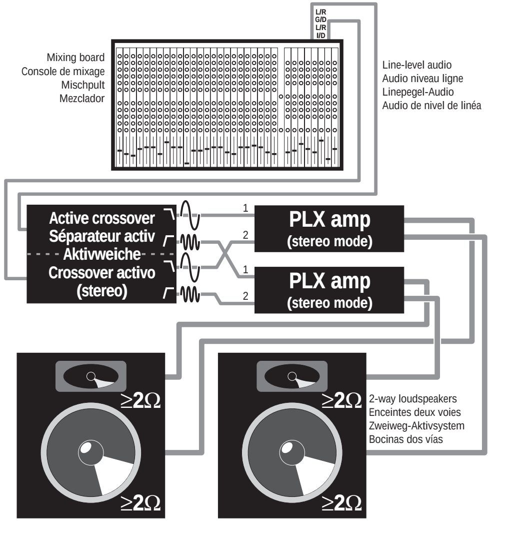
Stereo biamp mix #1


STEREO
L/R
G/D
L/R
I/D
Mixing board
Console de mixage
Mischpult
Mezclador
Line-level audio
Audio niveau ligne
Linepegel-Audio
Audio de nivel de linéa
Active crossover
Séparateur activ
Aktivweiche
Crossover activo
(stereo)
Speakon
1
2
PLX amp
(stereo mode)
1
2
PLX amp
(stereo mode)
Ch 2
Ch 1
Ch 2
Ch 1
≥2Ω
≥2Ω
≥2Ω
≥2Ω
2-way loudspeakers
Enceintes deux voies
Zweiweg-Aktivsystem
Bocinas dos vías

Biamp monitor


Monitor mix
Mixing board
Console de mixage
Mischpult
Mezclador
Line-level audio
Audio niveau ligne
Linepegel-Audio
Audio de nivel de linéa
Active crossover
Séparateur activ
Aktivweiche
Crossover activo
1
2
PLX amp
(stereo mode)
≥2Ω
≥2Ω
2-way monitor
Enceinte monitor deux voies
Zweiweg-Aktivmonitorsystem
Bocina monitor dos vías

Stereo biamp mix #2


STEREO
L/R
G/D
L/R
I/D
Mixing board
Console de mixage
Mischpult
Mezclador
Line-level audio
Audio niveau ligne
Linepegel-Audio
Audio de nivel de linéa
Active crossover
Séparateur activ
Aktivweiche
Crossover activo
(stereo)
1
2
PLX amp
(stereo mode)
1
2
PLX amp
(stereo mode)
≥2Ω
≥2Ω
≥2Ω
≥2Ω
2-way loudspeakers
Enceintes deux voies
Zweiweg-Aktivsystem
Bocinas dos vías

APPLICATIONS

Instrument amplification

APPLICATIONS

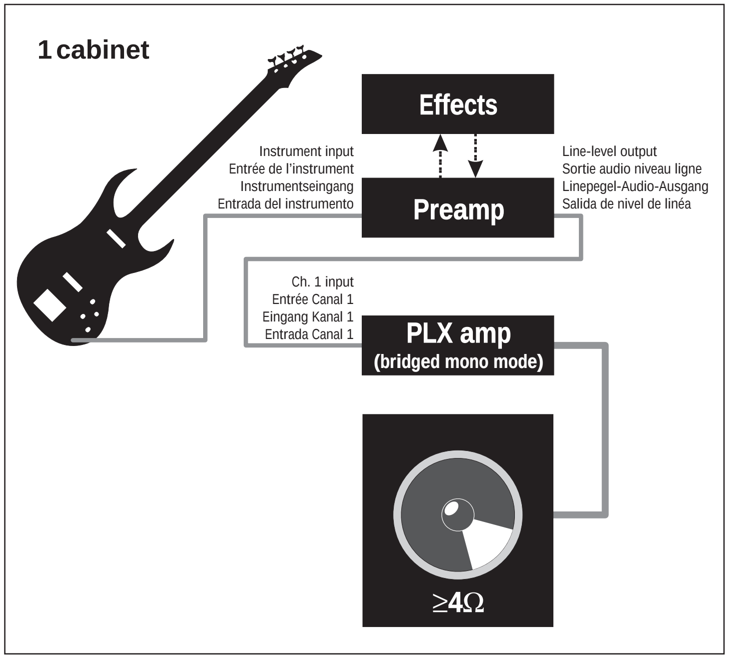
Amplification d'instrument

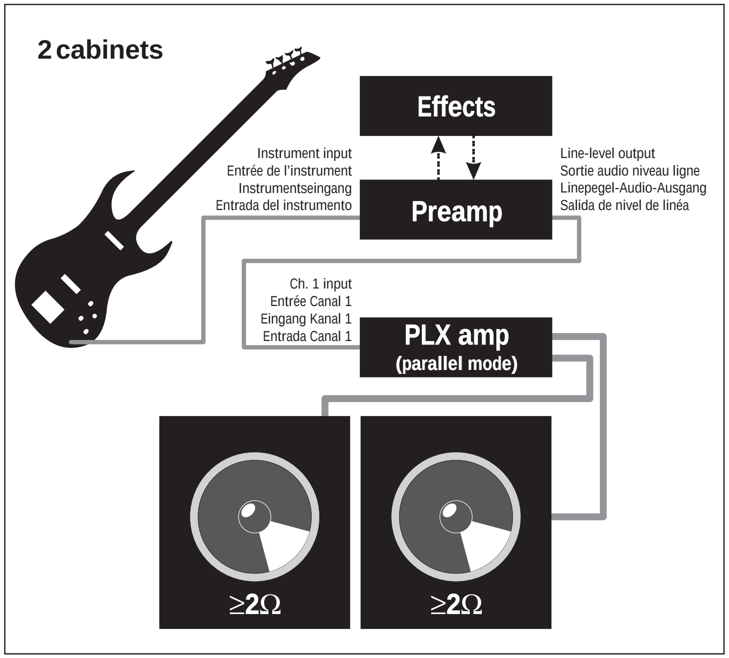
ANWENDUNGSBEISPIELE

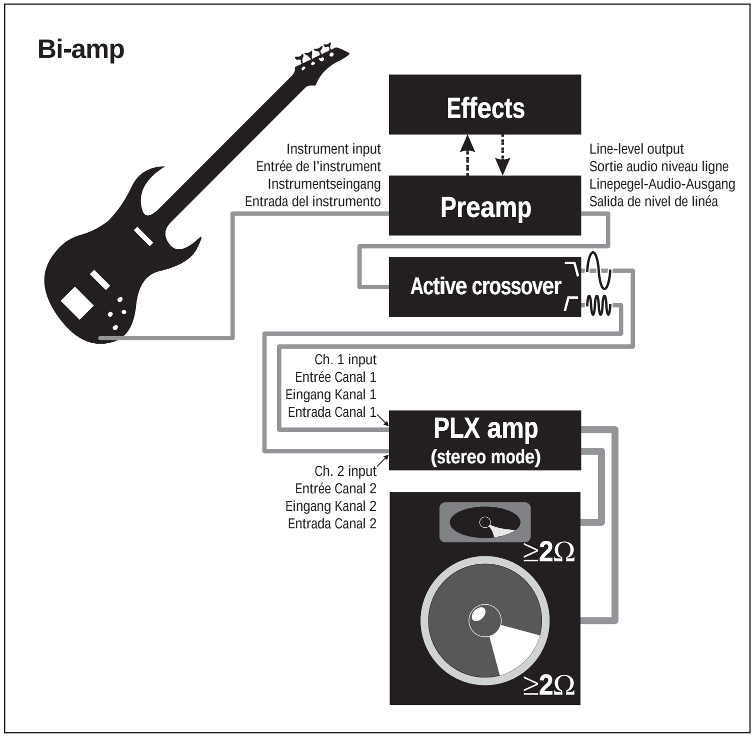
Instrument-Verstärkung

APLICACIÓNES

Amplifación de instrumento


1 cabinet
Effects
Preamp
Instrument input
Entrée de l'instrument
Instrumentseingang
Entrada del instrumento
Line-level output
Sortie audio niveau ligne
Linepegel-Audio-Ausgang
Salida de nivel de linéa
Ch. 1 input
Entrée Canal 1
Eingang Kanal 1
Entrada Canal 1
PLX amp
(bridged mono mode)
≥4Ω


2 cabinets
Effects
Preamp
Instrument input
Entrée de l'instrument
Instrumentseingang
Entrada del instrumento
Line-level output
Sortie audio niveau ligne
Linepegel-Audio-Ausgang
Salida de nivel de linéa
Ch. 1 input
Entrée Canal 1
Eingang Kanal 1
Entrada Canal 1
PLX amp
(parallel mode)
≥2Ω
≥2Ω

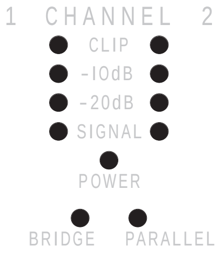

Bi-amp
Effects
Preamp
Instrument input
Entrée de l'instrument
Instrumentseingang
Entrada del instrumento
Line-level output
Sortie audio niveau ligne
Linepegel-Audio-Ausgang
Salida de nivel de linéa
Active crossover
Ch. 1 input
Entrée Canal 1
Eingang Kanal 1
Entrada Canal 1
PLX amp
(stereo mode)
Ch. 2 input
Entrée Canal 2
Eingang Kanal 2
Entrada Canal 2
≥2Ω
≥2Ω

Key • Légende
Zeicherklärung • Leyende

☼ = **lit**
allumé
aufgeleuchtet
illuminado

◐ = **blinking**
clignotant
blinkt
parpadeo

● = **off**
éteint
aus
apagado

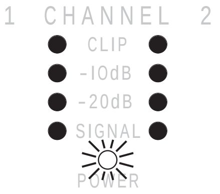


TROUBLESHOOTING

Problem: no sound

➤ ***INDICATION:*** **POWER INDICATOR NOT LIT**

- Check the AC plug.
- Confirm that the AC outlet works by plugging in another device. If too many amplifiers are used on one outlet, the building's circuit breaker may trip and shut off power.
- An overload in *bridged mono* mode may cause the amplifier to click off for three seconds, indicated by the half-bright ***POWER*** LED, followed by a normal restart cycle. Check the load impedance (4 ohms minimum), or reduce signal level. ***CLIP*** LEDs glowing bright red indicate a thermal shutdown.
- An amplifier which keeps shutting off may have a serious internal fault. Turn it off, remove AC power, and have the amplifier serviced by a qualified technician.



➤ ***INDICATION:*** **SIGNAL LED NOT LIT**

- If the green ***POWER*** indicator LED is at full brightness and the fan is running, yet the signal LEDs indicate no signal, check the input. Make sure the signal source is operating and try another input cable. Connect the source to another channel or amplifier to confirm its operation.

DÉPANNAGE

Problème: pas de son

➤ ***INDICATION:*** **DEL "POWER" ÉTEINTE**

- Vérifiez le branchement du cordon d'alimentation.
- Assurez vous que la prise de courant fonctionne en y branchant un autre appareil. Un trop grand nombre d'amplificateur reliés au même circuit pourraient faire déclencher le disjoncteur, coupant l'alimentation.
- Une surcharge en mode ponté mono pourrait éteindre momentanément l'amplificateur, tel qu'indiqué par la DEL ***POWER*** qui passera à demi-intensité, suivi par le cycle normal d'allumage. Vérifiez l'impédance de la charge (4 ohms minimum), ou réduisez le niveau de signal. L'allumage continu des DEL ***CLIP*** indique la mise en sourdine par le circuit de protection à cause de surchauffe.
- Un amplificateur qui passe continuellement en mode protection pourrait être défectueux. Eteignez l'amplificateur, débranchez le, puis faites le vérifier par un technicien compétent.

➤ ***INDICATION:*** **DEL SIGNAL ÉTEINTS CHECK**

- Si la DEL ***POWER*** est allumée et que le ventilateur fonctionne, mais que les DEL ***SIGNAL*** restent éteintes, vérifiez la source de signal. Assurez vous que la source fonctionne normalement, puis essayez d'autres câbles de raccord. Brancher le signal de source à un autre canal d'amplificateur pour confirmer la présence de signal.

FEHLERBEHEBUNG

Problem: kein Ton

➤ ***ANZEICHEN:*** **POWER ANZEIGE LEUCHTET NICHT**

- Überprüfen Sie den Netzstecker.
- Stellen Sie sicher, daß die Steckdose Strom führt, indem Sie ein anderes Gerät einstecken. Falls zu viele Verstärker angeschlossen sind, kann die Haussicherung ansprechen und die Versorgung unterbrechen.
- Eine Übersteuerung im Monobrückenbetrieb kann den Verstärker für drei Sekunden stummschalten, was durch eine halbhelle ***POWER*** LED angezeigt wird. Überprüfen Sie die Impedanz der angeschlossenen Lautsprecher (minimal 4Ω!), oder reduzieren Sie den Signalpegel. Eine hell aufleuchtende, rote ***CLIP*** LED zeigen eine thermische Abschaltung an.
- Ein Verstärker der ausgeschaltet bleibt, kann einen internen Defekt haben. Schalten Sie ihn aus, trennen Sie ihn von der Netzversorgung und lassen Sie den Verstärker von qualifiziertem Servicepersonal überprüfen.

➤ ***ANZEICHEN:*** **SIGNAL ANZEIGEN LEUCHTEN NICHT**

- Wenn die grüne ***POWER*** LED in voller Helligkeit leuchtet, der Lüfter läuft, die Signal LEDs aber kein Signal anzeigen, überprüfen Sie den Eingang. Stellen Sie sicher, daß die Quelle arbeitet und versuchen Sie es mit einem anderen Eingangskabel. Schließen Sie die Quelle an einen anderen Verstärker an, um sicherzustellen, daß sie arbeitet.

BUSQUEDA DE AVERÍAS

Problema: no hay sonido

➤ **INDICACIÓN:** ***EL INDICADOR DE ENCENDIDO NO SE ILUMINA***

- Revise el cable de corriente.
- Confirme que la toma de corriente funcione conectando otro aparato. Si se conectan muchos amplificadores a una sola toma, el circuito se sobrecarga y se desconecta.
- Una sobrecarga en el modo de puenteado en mono puede causar un apagado momentáneo, como de tres segundos, y el indicador LED de ***POWER*** bajará la intensidad de su luz, reiniciando la secuencia de encendido. Verifique la impedancia de carga (4 ohmios mínimo), o reduzca el nivel de la señal. Si el LED rojo de ***CLIP*** se ilumina indicará un apagado térmico.
- Un amplificador que se apagada continuamente, puede tener una falla interna. Apáguelo, desconéctelo de la toma de corriente y llévelo con un técnico especializado para su revisión.

➤ ***INDICACIÓN:*** **EL INDICADOR LED DE SEÑAL NO SE ILUMINA**

- Si el indicador verde de ***POWER*** está completamente iluminado, y el ventilador está funcionando, pero el LED de signal no indica nada, revise las entradas. Asegúrese de que exista la señal y cambie los cables en las entradas. Conecte la señal a otro canal u otro amplificador para confirmar su funcionamiento.

Key • Légende
Zeicherklärung • Leyende

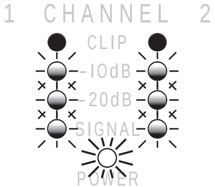
☼ = **lit**
allumé
aufgeleuchtet
illuminado

◌ = **blinking**
clignotant
blinkt
parpadeo

● = **off**
éteint
aus
apagado

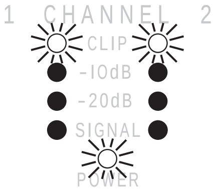
TROUBLESHOOTING

No sound (continued)



➤ *INDICATION:* SIGNAL LEDs RESPONDING TO SIGNAL LEVEL

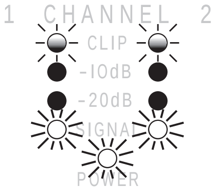
- If the green **SIGNAL, -20 dB,** and **-10 dB** indicators are lighting normally, the fault is somewhere between the amp and the speaker. Check the speaker wiring for breaks. Try another speaker and cable.



➤ *INDICATION:* CLIP LEDs BRIGHT AND STEADY

The amplifier is in protective muting.

- One second of muting is normal when the amp is turned on or off.
- Overheating will cause protective muting. The fan will be running at full speed and the chassis will be hot to the touch; sound should resume within a minute as the amplifier cools to a safe operating temperature. Check for proper ventilation. If the fan isn't running at all, the amplifier requires servicing.



➤ *INDICATION:* CLIP LED FLASHING

- If the red ***CLIP*** indicator flashes when signal is applied, the amplifier output may be shorted. Check the speaker wiring for stray strands or breaks in the insulation.

DÉPANNAGE

Pas de son (suite)

➤ *INDICATION:* DEL DE SIGNAL FONCTIONNENT NORMALEMENT

- Si les DEL ***SIGNAL***, ***-20dB*** et ***-10dB*** fonctionnent normalement, le problème se situe entre la sortie de l'amplificateur et le haut-parleur. Vérifiez l'intégrité du câble de haut-parleur. Essayez un autre câble et un autre haut-parleur.

➤ *INDICATION:* DEL CLIP ALLUMÉES EN CONTINU

L'amplificateur est en mode protection.

- La mise en sourdine pour une seconde est normale lors de la mise en tension et hors-tension de l'amplificateur.
- La surchauffe entraîne la mise en sourdine de l'amplificateur. Le ventilateur tournera alors à pleine vitesse et le châssis sera chaud au contact; le son devrait revenir sitôt l'amplificateur refroidi à température normale d'utilisation. Assurez une ventilation suffisante. Si le ventilateur ne tournait pas, l'amplificateur serait défectueux.

➤ *INDICATION:* DEL CLIP CLIGNOTE

- Si la DEL ***CLIP*** clignote lors de la présence de signal, la sortie de l'amplificateur est court-circuitée. Vérifiez l'intégrité du câblage.

FEHLERBEHEBUNG

Kein Ton (Fortsetzung)

➤ *ANZEICHEN:* SIGNAL ANZEIGEN LEUCHTEN NORMAL

- Wenn die grüne ***SIGNAL***, ***-20dB*** und ***-10dB*** LEDs normal aufleuchten, liegt der Fehler an anderer Stelle zwischen Verstärker und Lautsprecher. Überprüfen Sie die Lautsprecherleitungen auf Unterbrechungen. Schließen Sie einen anderen Lautsprecher mit anderen Kabeln an.

➤ *ANZEICHEN:* CLIP ANZEIGE LEUCHTET HELL UND GLEICHMÄßIG

Die Schutzschaltung hat den Verstärker abgeschaltet.

- Eine Abschaltung von einer Sekunde Dauer ist beim Ein- oder Ausschalten normal.
- Überhitzung führt zu einer Abschaltung des Verstärkers. Der Lüfter läuft mit höchster Geschwindigkeit und das Chassis fühlt sich heiß an. Der Ton sollte innerhalb einer Minute zurückkehren, sobald der Verstärker wieder eine normale Betriebstemperatur erreicht hat. Überprüfen Sie, ob eine einwandfreie Kühlung möglich ist. Falls der Lüfter nicht läuft, muß der Verstärker zur Reparatur eingeschickt werden.

➤ *ANZEICHEN:* CLIP LED BLINKT

- Wenn die rote ***CLIP*** LED bei anliegendem Eingangssignal blinkt, kann der Verstärkerausgang kurzgeschlossen sein. Überprüfen Sie die Anschlußklemmen auf eventuell heraussstehende Drähte oder Brüche der Isolation.

BUSQUEDA DE AVERÍAS

No hay sonido (continuación)

➤ *INDICACIÓN:* EL INDICADOR LED DE "SIGNAL" Y LOS INDICADORES DE NIVELES SE ENCIENDAN AL DETECTAR EL NIVEL DE LA SEÑAL

- Si los indicadores LED verdes de ***SIGNAL***, ***-20 dB*** y ***-10 dB*** parpadean, la falla ocurre entre el amplificador y las bocinas. Revise los cables de las bocinas, e intente cambiarlas o cambiar los cables.

➤ *INDICACIÓN:* LOS INDICADORES CLIP PERMANECEN ENCENDIDOS

El amplificador está haciendo un "enmudecimiento" (*mute*) de protección.

- Un segundo de asilenciamiento (*mute*) es normal cuando se enciende o se apaga el amplificador.
- El sobrecalentamiento puede causar "enmudecimiento" de protección. El ventilador estará funcionando a su máxima velocidad y el armazón o caja del amplificador estará caliente al tacto. El sonido se suspenderá por un minuto mientras el amplificador vuelve a una temperatura adecuada de operación. Revise si existe la ventilación necesaria. Si el ventilador no se mueve, el amplificador necesita mantenimiento.

➤ *INDICACIÓN:* EL INDICADOR LED DE CLIP PARPADEA

- Si ésto sucede cuando entra una señal, la salida del amplificador puede estar en corto circuito. Vaya a los cables de las bocinas y revise si tienen alambres sueltos o averías en el aislamiento.

Key • Légende
Zeicherklärung • Leyende

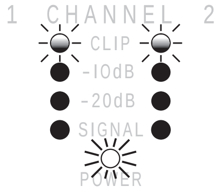
☼ = **lit**
allumé
aufgeleuchtet
illuminado

◎ = **blinking**
clignotant
blinkt
parpadeo

● = **off**
éteint
aus
apagado

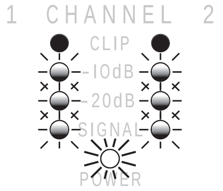
TROUBLESHOOTING

Problem: distorted sound



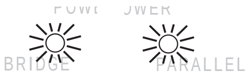
➤ *INDICATION:* CLIP LED FLASHING

- If the red ***CLIP*** indicator flashes before all three signal indicators do, the load impedance is abnormally low or shorted. Unplug each speaker one-by-one *at the amplifier.* If the ***CLIP*** LED goes out when you disconnect a cable, that cable or speaker is shorted. Try another cable and speaker to locate the fault.



➤ *INDICATION:* CLIP INDICATOR NOT FLASHING

- This could be caused by a faulty speaker or loose connection. Check the wiring and try another speaker.
- The signal source may be clipping. Keep the amplifier gain controls at least halfway up so that the source does not have to be overdriven.



Problem: no channel separation

- Check the yellow **PARALLEL** or **BRIDGE MONO** LEDs on the front panel, which indicate the switch settings on the back of the amplifier. Neither should be lit in dual-channel, bi-amp, or stereo use where different signals go to each channel. Make sure the "Parallel Input" and "Bridge Mode" switches are *OFF*.
- Make sure other equipment in the signal path, such as mixers, preamps, etc., are set for stereo, not mono.

DÉPANNAGE

Problème: son distortionné

➤ *INDICATION:* DEL CLIP CLIGNOTE

- Si la DEL ***CLIP*** clignote avant que les trois DEL de niveau de signal, ou l'impédance de charge est trop faible, ou la sortie de l'amplificateur est court-circuitée. Débranchez les haut-parleurs un à un à l'amplificateur. Si la DEL ***CLIP*** s'éteint quand vous débranchez un câble, ou le câble ou le haut-parleur est court-circuité. Essayez un autre câble puis un autre haut-parleur pour localiser le problème.

➤ *INDICATION:* DEL CLIP N'ALLUMENT PAS

- Peut être dû à du câblage ou un haut-parleur défectueux. Vérifiez le câblage et essayez un autre haut-parleur.
- Le signal pourrait être distortionné à la source. Utilisez l'amplificateur avec les contrôles de gain au moins à mi-course afin d'éviter avoir à surcharger la source de signal.

Problème: pas de séparation des canaux

- Assurez vous que les DEL jaunes ***PARALLEL*** et ***BRIDGE MONO*** sont éteintes sur le panneau avant, ce qui indiquerait la position des interrupteurs sur le panneau arrière. Ni l'une ni l'autre ne devrait être allumée pour utilisation en mode stéréo, bi-amplification, ou deux canaux, quand des signaux distincts sont amenés aux deux canaux de l'amplificateur. Assurez vous que les interrupteurs *PARALLEL* et *BRIDGE* sont en position OFF.
- Assurez vous que les autres équipements dans la chaîne audio en amont, soit les pré-amplificateurs, mixers ou autres, sont réglés en mode stéréo, et non en mono.

FEHLERBEHEBUNG

Problem: Verzerrungen

➤ *ANZEICHE:* CLIP LED BLINKT

- Falls die rote ***CLIP*** LED blinkt, bevor alle drei Signal-Anzeigen aufleuchten, ist die angeschlossene Last entweder zu niedrig oder kurzgeschlossen. Schließen Sie jeden Lautsprecher, einen nach dem anderen, auf der Verstärkerseite ab. Wenn die ***CLIP*** LED erlischt, wenn ein Kabel abgeschlossen wird, weisen entweder Kabel oder Lautsprecher einen Kurzschluß auf. Versuchen Sie ein anderes Kabel/Lautsprecher um den Fehler zu finden.

➤ *ANZEICHE:* CLIP ANZEIGE BLINKT NICHT

- Hier liegt die Ursache wahrscheinlich bei einem fehlerhaften Lautsprecher oder einem Wackelkontakt. Überprüfen Sie die Verkabelung und versuchen Sie einen anderen Lautsprecher.
- Die Signalquelle kann verzerren. Halten Sie die Gain-Regler etwa halb aufgedreht, damit die Quelle nicht übersteuert werden muß.

Problem: keine Kanaltrennung

- Überprüfen Sie die frontseitigen, gelben ***PARALLEL*** oder ***BRIDGE MONO*** LEDs, welche die Schalterstellung auf der Rückseite anzeigen. Keine sollte im Zweikanal-, Biamp oder Stereobetrieb aufleuchten, bei denen unterschiedliche Signale den individuellen Kanälen zugeführt werden. Stellen Sie sicher, daß die "Parallel Input" und "Bridge Mono" Schalter auf *OFF* stehen.
- Stellen Sie sicher, daß andere Geräte der Signalkette, z.B. Mixer o.ä. nicht im Monobetrieb laufen.

BUSQUEDA DE AVERÍAS

Problema: sonido distorsionado

➤ *INDICACIÓN:* EL INDICADOR LED DE CLIP PARPADEA

- Si el indicador rojo de ***CLIP*** parpadea antes que los otros tres indicadores de señal, la impedancia de carga es anormal, muy baja o corta. Desconecte cada bocina, una por una, desde el amplificador. Si el indicador se apaga cuando usted desconecta un cable, éste o esa bocina estan bajos. Pruebe con otros para localizar la falla.

➤ *INDICACIÓN:* EL INDICADOR CLIP NO PARPADEA

- Esto puede ser causado por una defectuosa o por desconexión. Revise el cableado y pruebe con otra bocina.
- La fuente de la señal puede que se esté saturando. Mantenga los controles de ganancia del amplificador a más de medio trayecto, para que de esa manera no tenga que exigir demasiado al fuente de la señal.

Problema: no hay separación entre los canales

- Revise los indicadores amarillos ***PARALLEL*** o ***BRIDGE MONO*** en el panel frontal del amplificador, que indican la posición de los interruptores en la parte de atrás del amplificador. Ninguno debe iluminar si usan los modos de dos canales, biamplificado o estéreo. Revise que los interruptores "Parallel Input" y "Bridge mode" estén apagados.
- Verifique el resto del equipo en el trayecto de la señal, como mezcladores, preamplificadores, etc., estén ajustados en estéreo y no en mono.

TROUBLESHOOTING

Problem: hum

- The PowerWave™ supply eliminates internal hum fields, but AC transformers in other devices may cause hum. Move cabling and signal sources to identify "hot spots" in the system. Cables with faulty shielding are a frequent entry point for hum.

Problem: hiss

- Unplug the amplifier input to confirm that the hiss is coming from the source or a device upstream; erratic or popping noises indicate an electronic fault in the offending unit.
- To keep the normal noise floor low, operate the primary signal source at full level, without clipping, and avoid boosting the signal further between the source and the amplifier.

Problem: squeals and feedback

- Microphone feedback should be controlled with mixer controls. If noise continues to build up with zero mic gain, there is a serious fault in the signal processors or cables. Working in succession from the signal source towards the amplifier, check each device in the signal path by reducing its gain or unplugging it.

DÉPANNAGE

Problème: ronflement

- Le bloc d'alimentation PowerWave élimine les champs magnétiques internes, mais il est possible que les blocs d'alimentations d'autres appareils génèrent du bruit sous forme de ronflement (hum). Déplacez les câbles et/ou les appareils du système afin de localiser les sources de bruit. Les câbles avec un blindage défectueux sont une source commune de ronflement.

Problème: souffle

- Débranchez le câble d'entrée pour confirmer que le bruit provient d'autres équipements dans la chaîne audio en amont, soit les pré-amplificateurs, mixers ou autres; des bruits transitoires pourraient indiquer un problème électronique dans l'unité en cause.
- Afin de reduire le niveau de bruit de fond au minimum, utilisez les sources à un niveau de signal élevé, sans distortion ou écrêtage, et ajustez le niveau final sur l'amplificateur.

Problème: bruits et effet Larsen

- L'effet Larsen (feedback) par les microphones devrait être contrôlé sur le mixer. Si le problème persiste avec les micros fermés, il existe une problème dans le système, soit au niveau électronique ou au niveau du câblage. En travaillant à partir de la source, vérifiez chaque appareil de la chaîne en en réduisant le gain ou en le mettant hors-circuit.

FEHLERBEHEBUNG

Problem: Brummen

- Das PowerWave Netzteil eliminiert interne Brummfelder, aber Netztransformatoren in anderen Geräte können immer noch Brummen verursachen. Bewegen Sie die Verkabelung und Eingangsquellen hin und her, um Fehlerquellen im System aufzuspüren. Kabel mit fehlerhafter Abschirmung sind die häufigste Ursache für Brummschleifen.

Problem: Rauschen/ Zischen

- Trennen Sie den Verstärkereingang von der Quelle, um festzustellen, ob die Störgeräusche von einem vorgeschalteten Gerät stammen. Harte und poppende Störgeräusche lassen auf einen elektronischen Fehler im störenden Gerät schließen.
- Um den normalen Störgeräuschteppich niedrig zu halten, empfiehlt es sich, die Quelle mit vollem Pegel, aber ohne Übersteuerungen zu betreiben. Achten Sie auch darauf, den Signalpegel zwischen Quelle und Verstärker weiter anzuheben.

Problem: Pfeifen und Rückkopplungen

- Rückkopplungen die von Mikrofonen verursacht werden, sollten mit Hilfe eines Mischpultes unterdrückt werden. Wenn die Störgeräusche trotz ausgeschalteter Mikrofonverstärkung erhalten bleiben, liegt wahrscheinlich ein Fehler innerhalb eines Signalprozessors oder der Verkabelung vor. Eine schrittweise Überprüfung aller beteiligten Geräte, von der Signalquelle bis hin zum Verstärker, mit Überprüfung der jeweiligen Verstärkung, ist notwendig.

BUSQUEDA DE AVERÍAS

Problema: zumbidos

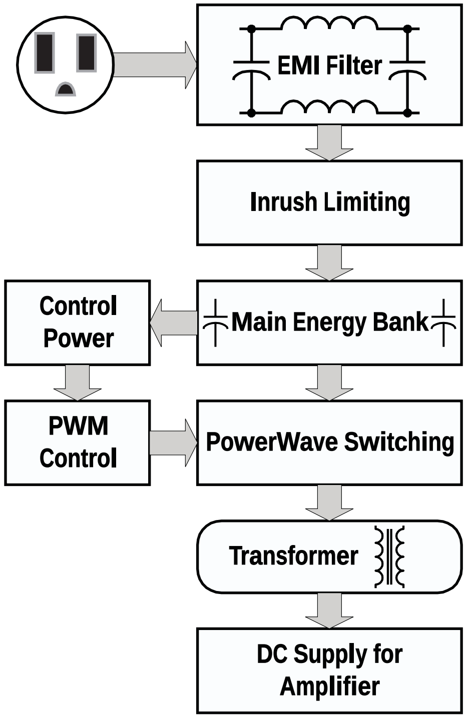
- La fuente de alimentación PowerWave™ elimina los zumbidos internos, pero los transformadores de corriente alterna (AC) de otros aparatos pueden causarlos. Mueva los cables y la fuente de la señal para encontrar los "puntos débiles" del sistema. Los cables con aislamiento defectuoso por lo general son un punto de entrada de zumbidos y ruidos.

Problema: ruido *(hiss)*

- Desconecte la entrada del amplificador para revisar si el ruido proviene desde la fuente de la señal o de otro aparato intermedio. El ruido permanente o explosivo revela una falla eléctrica en la unidad.
- Para mantener el ruido de piso normal abajo, mantenga la fuente de la señal a un volumen máximo, sin saturación, y absténgase de resaltar la señal entre la fuente y el amplificador.

Problema: chillidos y retroalimentación

- La retroalimentación del micrófono puede controlarse desde el mezclador. Si el ruido continua, aún con la ganancia del micrófono en cero, entonces debe de haber una falla seris en los procesadores de señales o en los cables. Siga el trayecto de la señal y desconecte o baje el volumen de los aparatos uno por uno hacia el amplificador.



INNER WORKINGS

An impressive amount of technology is packed "under the hood" of a PLX Series amplifier. Thousands of watts of power flow inches away from state-of-the-art low noise inputs. Precise circuit layout and thorough protection assure that all of this activity occurs smoothly and safely. So, what actually happens when you turn on the power switch?

Soft Start Sequence. The first task is to charge the primary energy reservoir without drawing a large surge current. A special inrush resistor allows just enough current to charge the energy bank in three seconds. Meanwhile, a low-power switching supply provides power to start up the main supply. After three seconds, a relay bypasses the inrush resistor and full power operation is enabled. The audio circuitry mutes for one second to eliminate start-up thumps. When the red ***CLIP*** lights go out, the amplifier is ready for action.

PowerWave™ Technology. High current switching devices draw over 10,000 watts of peak power from the main energy reservoir, which is replenished directly from the AC line for maximum stiffness. Conventional amplifiers must isolate the energy bank with a large AC transformer, which weakens the flow of current, allows greater sag under load, and produces hum. The PowerWave supply performs voltage conversion at a very high frequency, allowing better coupling through a much smaller isolation transformer.

SOUS LE CAPOT

Une quantité impressionnante de technologie se trouve "sous le capot" des amplificateurs de série PLX de QSC. Des milliers de watts de puissance déferlent à la sortie de l'amplificateur, à quelques centimètres à peine des circuits d'entrée à faible bruit et de technologie de pointe. L'implantation précise des composantes électroniques et les circuits de protections sophistiqués assurent le fonctionnement sécuritaire et sans soubresaut. En fait, qu'arrive-t'il quand vous mettez l'amplificateur en marche?

Séquence Soft Start. La première tâche de l'amplificateur est de charger le réservoir d'énergie primaire sans faire un appel de courant trop élevé. Une résistance spéciale est installée de façon à ce que l'amplificateur prenne trois secondes pour charger le réservoir d'énergie. Pendant ce temps, un circuit d'alimentation secondaire à faible courant s'occupe de mettre en marche le circuit d'alimentation à commutation principal. Une fois les trois secondes écoulées, un relais dérive la résistance d'appel de courant, et l'amplificateur fonctionne normalement à pleine puissance potentielle. Le circuit de protection met le circuit audio en sourdine pendant une seconde pour éviter les à-coups de démarrage. Quand la DEL ***CLIP*** s'éteint, l'amplificateur est prêt à fonctionner.

La technologie PowerWave. Des dispositifs de commutation à courant élevé peuvent passer plus de 10,000 watts en puissance de pointe à partir du réservoir d'énergie principal, qui est lui-même directement alimenté par la source de courant CA, pour une tension de base très stable. Les amplificateurs conventionnels doivent isoler le réservoir d'énergie de la source de courant avec un transformateur de grande dimension, ce qui réduit le transfert de courant, permet une chute de tension en pointe de courant, et produit du ronflement. Le bloc d'alimentation à commutation PowerWave effectue le conversion de tension à très haute fréquence, ce qui rend possible un bien meilleur couplage à travers un petit transformateur.

INNERER AUFBAU

In jeder PLX ist beeindruckende Technologie verborgen. Tausende Watt Leistung fließen nur Zentimeter entfernt von State-of-the-Art rauscharmen Eingängen. Ein präziser Schaltungsaufbau und gründliche Abschirmungen garantieren einen problemlosen und sicheren Betrieb. Daher interessiert es Sie vielleicht, was nach dem Einschaltvorgang im Verstärker passiert.

Softstart Sequenz. Die erste Aufgabe hierbei ist, das primäre Energiereservoir aufzuladen, ohne einen zu hohen Einschaltstrom zu verursachen. Ein spezieller Einschaltwiderstand ermöglicht ausreichend Energie innerhalb von drei Sekunden an die Netzkondensatoren zu liefern. In der Zwischenzeit stellt ein schwaches, getaktetes Netzteil die Energie zur Verfügung, um das Hauptnetzteil zu starten. Nach diesen drei Sekunden , überbrückt ein Relais den Einschaltwiderstand und die volle Energie steht zur Verfügung. Die Audio-Schaltkreise schalten für eine Sekunde stumm, um Einschaltknackgeräusche zu unterdrücken. Wenn die roten ***CLIP*** LEDs erlöschen ist der Verstärker betriebsbereit.

PowerWave Technologie. Hochleistungs-Schaltnetzteile ziehen über 10000 Watt Spitzenleistung aus den Netzkondensatoren, die direkt aus der Netzleitung versorgt werden. Konventionelle Verstärker müssen diese Kondensatoren mit Hilfe eines großen Transformators versorgen, was zu höheren Einbrüchen bei hohem Strombedarf führt, den zur Verfügung stehenden Strom begrenzt und zusätzlich Brummen verursacht. Die PowerWave Technologie wandelt die Spannungen bei sehr hoher Frequenz, was mit wesentlich kleineren Transformatoren zu einer besseren Ausnützung führt.

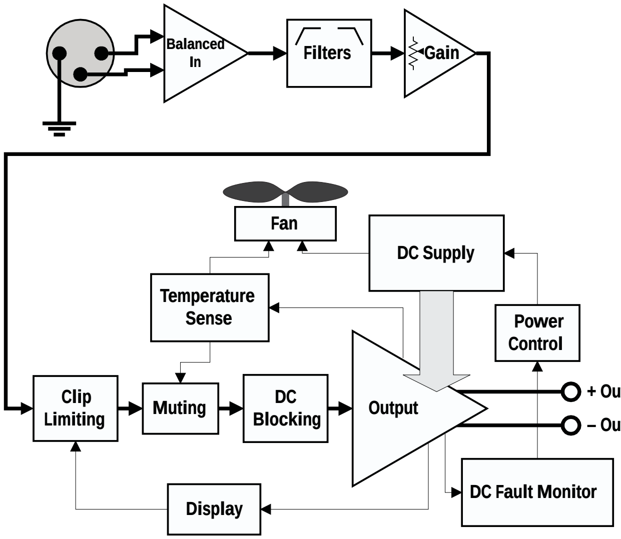
OPERACIÓN INTERNA

Una impresionante tecnología se encuentra "escondida" en los amplificadores de la Serie PLX. Miles de wattios de potencia se mueven a pocos centímetros de sus entradas de bajo ruido.

Los elaborados diagramas de circuito y una protección general le dan la confianza de que toda esa actividad es precisa y segura. Pero, ¿qué es lo que realmente ocurre cuando usted presiona el interruptor de encendido?

Secuencia de encendido suave. La primera tarea es cargar el receptor primario de energía sin una gran cantidad de corriente. Una resistencia de entrada permite tomar sólo la corriente necesaria para cargar el banco de energía en tres segundos. Mientras tanto un interruptor de baja potencia enciende la fuente principal. Después de tres segundos, se bloquea la resistencia de entrada y puede así operar a su máxima potencia. El circuito de audio se asilencia por un segundo para eliminar cualquier ruido de encendido Cuando la luz roja de ***CLIP*** se apaga, quiere decir que el amplificador está listo para usarse.

La tecnología PowerWave™. Los aparatos con interruptores de corriente alta "jalan" más de 10,000 wattios de potencia en picos desde el contenedor principal de energía, el cual se alimenta directamente de la línea de corriente alterna (AC) para máxima confiabilidad. Los amplificadores convencionales deben aislar el banco de energía con grandes transformadores AC, lo que debilita el flujo de corriente, provocando una gran pérdida de carga, y produciendo zumbidos o *hum*. La fuente de alimentación PowerWave hace la conversión de voltaje a una frecuencia muy alta, acoplándose mejor a un transformador de aislamiento más pequeño.



INNER WORKINGS

High Performance Audio. High speed power transistors convert this DC power into the full range audio output which drives the speakers. High-current design and special dual-sense output feedback corrects errors on both sides of the speaker terminals, improving damping and control of speaker motion. The power devices are directly mounted to isolated heat sinks, which form a short, wide air tunnel in front of the fan for optimum cooling.

A thermal sensor embedded in each channel's heat sink monitors the temperature and controls fan speed, thermal shutdown, and bias control, assuring maximum audio clarity at all temperatures and signal levels. A circuit monitors transistor dissipation and triggers protective cutback only when actually needed.

The output circuitry is actively clamped during clipping for smooth and very fast recovery. The clamp also feeds a proportional clip limiter, which actually senses the depth of clipping and responds accordingly.

The balanced inputs use premium 0.1% precision resistors for very high noise rejection. The precision components used in the input filters and all other circuitry ensure accurate performance.

Shutdown. The amplifier mutes as soon as power is shut off, preventing turn-off noises. Serious faults trigger a shutdown of the power supply; the high switching frequency cuts off power within microseconds to limit damage.

SOUS LE CAPOT

Audio haute performance. Des transistors de puissance à haute vitesse convertissent cette puissance de courant continu en une sortie audio plein registre qui actionne le haut-parleur. La conception à haute tension et le circuit spécial de rétroaction à double détecteur corrige les erreurs sur les tensions positives et négatives des haut-parleurs, améliorant ainsi le coefficient d'amortissement et le contrôle du mouvement du cône du haut-parleur. Les composantes de puissance sont montées directement sur le radiateur isolé, qui lui-même forme un tunnel d'air large et court directement devant le ventilateur, pour un refroidissement optimal.

Une sonde thermique est noyée à même le radiateur de chaque canal pour contrôler la température et piloter la vitesse du ventilateur, le circuit de protection, le circuit de polarisation, assurant ainsi une performance audio maximale à tout niveau de température ou de signal. Un circuit supervise la dissipation des transistors et n'active le délestage de protection que lorsque vraiment nécessaire.

L'étage de sortie est bloqué électroniquement lors de l'écrêtage pour obtenir un rétablissement très doux et ultra rapide. Le circuit de blocage pilote aussi le circuit limiteur proportionnel, en détectant la montant d'écrêtement et réagissant en conséquence.

Les entrées symétriques utilisent des résistances de précision à 0.1% afin d'obtenir un facteur de différence (CMRR) très élevé. Les composantes de précision utilisées dans les filtres d'entrée et aux autres étages du circuit audio assurent une reproduction des plus fidèles.

Arrêt. L'amplificateur passe en sourdine aussitôt qu'on l'éteint, éliminant ainsi tout bruit ultérieur. Toute panne majeure déclenchera le bloc d'alimentation; le circuit de commutation à haute fréquence est désengagé en quelques microsecondes afin de limiter les dégâts.

INNERER AUFBAU

High Performance Audio. Hochgeschwindigkeitstransistoren wandeln diese Gleichspannungsleistung in Audiosignale mit voller Bandbreite um, welche zum Betrieb der Lautsprecher verwendet werden. Ein Hochleistungs-Schaltungsaufbau und zweifache Ausgangsüberprüfung korrigiert Fehler auf beiden Seiten der Lautsprecheranschlüsse, verbessert den Dämpfungsfaktor und steuert die Lautsprecherbewegungen. Die Leistungskomponenten sind direkt mit isolierten Kühlkörpern verbunden. Zur Erzielung optimaler Kühlung bilden diese Kühlkörper eine breite Luftschleuse direkt vor einem Lüfter. Ein Thermofühler im Kühlkörper jedes Kanals mißt die Temperatur und steuert Lüftergeschwindigkeit, thermisches Abschalten und die Steuerung des BIAS, wodurch höchste Audioqualität bei allen Betriebstemperaturen und Signalpegeln gewährleistet wird. Ein Schaltkreis überwacht die Stromaufnahme der Transistoren und löst schützende Reduktion der Leistung nur dann aus, wenn sie auch tatsächlich benötigt wird.

Der Ausgangskreis wird während Übersteuerungen aktiv reduziert, um eine weiche und schnelle Wiederherstellung des Signals zu ermöglichen. Diese Schutzschaltung steuert zusätzlich einen proportional arbeitenden Spitzenbegrenzer, der die tatsächliche Stärke der Übersteuerung mißt und entsprechend reagiert.

Die symmetrischen Eingänge verwenden hochqualitative 0,1% Eingangswiderstände für beste Störgeräuschunterdrückung. Die Präzisionsbauteile, die in den Eingangsfiltern Verwendung finden, ermöglichen eine akkuraten, problemlosen Betrieb.

Abschalten. Der Verstärker schaltet die Ausgangskreise stumm, sobald die Netzversorgung abgeschaltet wurde, um hierdurch Ausschaltgeräusche wirksam zu unterdrücken. Schwerwiegende Fehler verursachen ein Ausschalten des Netzteiles; die hohe Schaltgeschwindigkeit des Netzteils schaltet innerhalb weniger Mikrosekunden ab, um Schaden zu begrenzen.

OPERACIÓN INTERNA

Audio de alto rendimiento. Los transistores de potencia de alta velocidad, convierten esta energía DC en una salida de audio de rango completo que va a las bocinas. El diseño de corriente alta y una salida especial de doble sentido corrige errores en ambos lados de las terminales de las bocinas, mejorando la amortiguación y el control del movimiento de la misma. Los aparatos de potencia están colocados sobre superficies con aislamiento térmico, lo que forma un túnel de aire frente al ventilador para un enfriamiento óptimo.

Un sensor térmico puesto en cada canal monitorea la temperatura y controla la velocidad del ventilador, del apagado térmico y del control de polarización (bias), asegurando máxima claridad de audio a cualquier temperatura y nivel de señal. Un circuito monitorea la disipación de los transistores y realiza cortes de protección sólo cuando realmente se necesita.

El circuito de salida se comprime activamente durante la saturación para permitir una recuperación rápida y suave. El compresor también alimenta un limitador de picos proporcional, el cual realmente siente la intensidad del pico y responde de manera semejante.

Las entradas balanceadas usan resistores *premium* al 0.1% de precisión para un alto rechazo al ruido. Los componentes de precisión usados en los filtros de entrada y el resto de accesorios del circuito le aseguran un funcionamiento exacto.

Apagado. El amplificador asilencia tan pronto como se corta la corriente, evitando ruidos de apagado. El apagado de la fuente de alimentación puede provocar un daño serio. La frecuencia alta del interruptor corta la potencia en microsegundos para evitar un daño mayor.

SPECIFICATIONS | SPÉCIFICATIONS | TECHNISCHE DATEN | ESPECIFICACIONES

	PLX 1202	PLX 1602	PLX 2402	PLX 3002	PLX 3402
OUTPUT POWER *in watts*					
20 Hz–20 kHz @ 0.03% THD					
8Ω per channel	200	300	425	550	700
20 Hz–20 kHz @ 0.05% THD					
4Ω per channel	325	500	700	900	1100
EIA: 1 kHz @ 1% THD					
8Ω per channel	215	325	475	625	800
4Ω per channel	375	550	825	1050	1250
2Ω per channel	600	800	1200	1500	1700
Bridge Mono:					
16Ω, 20 Hz–20 kHz, 0.1% THD	400	600	850	1100	1400
8Ω, 20 Hz–20 kHz, 0.1% THD	700	1000	1500	2000	2200
4Ω, 1 kHz, 1% THD	1200	1600	2400	3000	3400
DYNAMIC HEADROOM	2 dB @ 4Ω				
DISTORTION					
SMPTE-IM	< 0.01%	< 0.01%	< 0.02%	< 0.02%	< 0.02%
FREQUENCY RESPONSE (at 10 dB below rated output power)	20 Hz–20 kHz, ±0.15 dB -3 dB points: 5 Hz and 100 kHz				
DAMPING FACTOR	> 500 @ 8Ω				
NOISE (unweighted 20 Hz to 20 kHz, below rated output)	106 dB	107 dB	108 dB	107 dB	107 dB
VOLTAGE GAIN	40× (32 dB)				
INPUT SENSITIVITY, V RMS					
full rated power @ 8Ω	1.0v (+2.2 dBu)	1.2v (+3.9 dBu)	1.5v (+5.5 dBu)	1.7v (+6.8 dBu)	1.9v (+7.7 dBu)
full rated power @ 4Ω	0.9v (+1.3 dBu)	1.1v (+3.2 dBu)	1.3v (+4.6 dBu)	1.5v (+5.7 dBu)	1.7v (+6.6 dBu)
INPUT IMPEDANCE	6 KΩ unbalanced 12 KΩ balanced				
CONTROLS	Front: AC switch, Ch. 1 and Ch. 2 gain knobs with 11 detents Rear: 10-position DIP switch				
INDICATORS	POWER: Green LED PARALLEL: Yellow LED BRIDGED: Yellow LED		CLIP: Red LED, 1 per channel -10 dB: Green LED, 1 per channel -20 dB: Green LED, 1 per channel SIGNAL: Green LED, 1 per channel		
CONNECTORS	Input: XLR and ¼" (6.3 mm) TRS, tip and pin 2 positive Output: “Touch-Proof” binding posts and Neutrik Speakon™				
COOLING	Continuously variable speed fan, back-to-front air flow				
AMPLIFIER PROTECTION	Full short circuit, open circuit, thermal, ultrasonic, and RF protection Stable into reactive or mismatched loads				

SPECIFICATIONS | SPÉCIFICATIONS | TECHNISCHE DATEN | ESPECIFICACIONES

	PLX 1202	PLX 1602	PLX 2402	PLX 3002	PLX 3402
LOAD PROTECTION	Turn-on/turn-off muting, DC-fault power supply shutdown, clip limiting				
OUTPUT CIRCUIT TYPE	AB	AB	H	H	H

AB: Class AB complementary linear output
H: Class AB complementary linear output with Class H 2-step high efficiency circuit

DIMENSIONS

19.0" (48.3 cm) wide, 3.5" (8.9 cm) tall (2 rack spaces)
13.2" (33.5 cm) deep (rack mounting to rear support ears)

WEIGHT

Shipping: 27 lbs (12.3 kg) Net: 21 lbs (9.5 kg)

POWER REQUIREMENTS

Available for 120 or 220–240 VAC, 50/60 Hz

POWER CONSUMPTION ***@ 120 VAC***
(both channels driven)

Multiply currents by 0.5 for 230V units

PLX 1202

	Typical[1]	Full[2]	Max[3]
Idle	0.8 A		
8Ω	3.8 A	5.4 A	8.4 A
4Ω	6.0 A	8.9 A	14.0 A
2Ω	9.6 A	14.3 A*	23 A*

PLX 1602

	Typical[1]	Full[2]	Max[3]
Idle	0.9 A		
8Ω	5.6 A	8.0 A	12.5 A
4Ω	9.0 A	13.3 A	21 A
2Ω	14 A	21 A*	34 A*

PLX 2402

	Typical[1]	Full[2]	Max[3]
Idle	0.9 A		
8Ω	5.0 A	8.4 A	15.8 A
4Ω	7.9 A	13.5 A	26 A
2Ω	11.8 A	22 A*	42 A*

PLX 3002

	Typical[1]	Full[2]	Max[3]
Idle	0.9 A		
8Ω	6.0 A	11 A	20 A
4Ω	9.5 A	17 A	33 A
2Ω	14 A	27 A*	50 A*

PLX 3402

	Typical[1]	Full[2]	Max[3]
Idle	0.9 A		
8Ω	7.6 A	13.1 A	25 A
4Ω	11.6 A	20 A	39 A
2Ω	16.6 A		

POWER CONSUMPTION NOTES

[1] 1/8 power with pink noise represents typical program with occasional clipping.

[2] 1/3 power with pink noise represents severe program with heavy clipping.

[3] Continuous sine wave at 1% clipping.

* Thermal or overcurrent cutback limits duration.

US patents pending

SPECIFICATIONS SUBJECT TO CHANGE WITHOUT NOTICE

WARRANTY INFORMATION

(USA only; see your dealer or distributor)

Disclaimer

QSC Audio Products, Inc. is not liable for any damage to speakers, amplifiers, or any other equipment that is caused by negligence or improper installation and/or use of the PLX amplifier.

Product Warranty

QSC guarantees the PLX to be free from defective material and/or workmanship for a period of three years from the date of sale, and will replace defective parts and repair malfunctioning products under this warranty when the defect occurs under normal installation and use—provided the unit is returned to our factory via prepaid transportation with a copy of the proof of purchase, i.e., sales receipt. This warranty provides that examination of the returned product must indicate, in our judgment, a manufacturing defect. This warranty does not extend to any product which has been subjected to misuse, neglect, accident, improper installation, or where the date code has been removed or defaced.

INFORMATIONS DE GARANTIE

(É-U seulement; consultez votre marchand ou distributeur)

Décharge

QSC Audio products, Inc. ne peut être tenu responsable de tout dommage à des haut-parleurs, amplificateurs, ou tout autre équipement qui pourrait être dû à de la négligence ou mauvaise installation et/ou utilisation d'un amplificateur PLX.

Garantie de produit

QSC garantit le produit PLX libre de défaut de pièce et/ou de fabrication, et ce pour une période de trois ans à partir de la date d'achat, et remplacera les pièces défectueuses et réparera le produit sous l'effet de cette garantie en autant que le produit est installé et utilisé de façon normale, et que le produit est retourné à notre usine port payé, accompagné d'une copie de la preuve d'achat, i.e. facture originale. Cette garantie est conditionnelle à ce qu'une inspection du produit retourné révèle, selon notre jugement, un défaut de fabrication. Cette garantie ne couvre pas les produits ayant subi abus, négligence, accident, installation incorrecte, ou dont le code de date a été enlevé ou rendu illisible.

GARANTIE-BEDINGUNGEN

(Nur USA; in anderen Ländern Ihren Fachhändler fragen.)

Haftungserklärung

QSC Audio Products, Inc. haftet nicht für Schäden an Lautsprechern, Verstärkern, oder anderen Geräten, die durch Fahrlässigkeit im Betrieb oder durch nachlässige Installation verursacht wurden.

Produktgarantie

QSC garantiert für die PLX-Verstärker einwandfreie Herstellung und Freiheit von Materialmängeln für die Dauer von drei Jahren nach Verkaufsdatum. Innerhalb dieser Zeit ersetzt QSC defekte Teile und repariert nicht funktionierende Komponenten /Produkte, wenn der Defekt unter normalen Betriebsumständen auftritt. Dies bei frachtfreiem Versand zum Hersteller, mit Kaufquittung. Der Garantiefall muss nach unserer Untersuchung und nach unserem Urteil, durch einen Herstellungsfehler ausgelöst worden sein. Eine weitergehende Haftung für Produkte, die missbräuchlich genutzt wurden, durch Fahrlässigkeit beschädigt worden sind, durch Unfall, durch unsachgemässe Installation, oder bei Entfernung des Datumscodes, schliessen wir aus.

INFORMACIÓN DE GARANTÍA

(EE. UU. solamente; consulte su comerciante o su distribuidor)

Atención

QSC Audio Products, Inc., no es responsable por daños a las bocinas, amplificadores o cualquier otro equipo que sea causado por negligencia o mala instalación o uso de los amplificadores PLX.

Garantía

QSC garantiza que el PLX estará libre de defectos en piezas o mano de obra por un período de tres años de la fecha de venta, y cambiará las partes que no funcionen y arreglará productos cubiertos por esta garantía mientras que el defecto surja bajo condiciones normales de uso y asumiendo que la unidad será enviada a nuestra fábrica vía transporte prepagado con una copia de la prueba de compra (ejemplo: recibo de venta). Esta garantía dependerá de una examinación del producto devuelto y deberá indicar, a nuestro juicio, un defecto de fabrica. Esta garantía no se extiende a ningún producto que ha sido sometido a uso fuera de nuestras recomendaciones, accidentes, instalación deficiente y si el código de la fecha ha sido enmendado o retirado.

ADDRESS & TELEPHONE INFORMATION

ADRESSE POSTALE ET NUMÉROS

ANSCHRIFT UND TELEFON-NUMMERN

DIRECCIÓN Y TELÉFONO

Mailing address / Adresse postale / Postanschrift / Dirección postal:

QSC Audio Products, Inc.
1675 MacArthur Boulevard
Costa Mesa, CA 92626-1468 USA

Telephone Numbers / Numéros de téléphone / Telefonnummern / Números de teléfono:

Main Number / Numéro principal / Hauptnummer / Número principal	+(714) 754-6175	
Sales Direct Line / Ligne directe ventes / Verkauf-Direkt / Línea directo ventas	+(714) 957-7100	
Sales & Marketing / Ventes & marketing / Verkauf u. Marketing / Ventas y marketing	(800) 854-4079	*(toll-free in U.S.A. only)* *(sans frais aux É-U seulement)* *(zollfrei nur beim USA)* *(sin costo en EE. UU. solamente)*
Customer Service / Service à la clientèle / Kundendienst / Servicio a la clientela	+(714) 957-7150	
	(800) 772-2834	*(toll-free in U.S.A. only)* *(sans frais aux É-U seulement)* *(zollfrei nur beim USA)* *(sin costo en EE. UU. solamente)*

Facsimile Numbers / Numéros de télécopieur / Telefaxnummern / Número de FAX:

Sales & Marketing FAX / Télécopie ventes & marketing / Telefax der Verkauf u. Marketing / FAX ventas y marketing	+(714) 754-6174
Customer Service FAX / Télécopie service à la clientèle / Kundendienst-Telefax / FAX servicio a la clientela	+(714) 754-6173

World Wide Web: **www.qscaudio.com**

BBS/World Group:

QSC OnLine Technical Support / Support technique QSC enligne / Technische fachberaten QSC „On-Line“ / Soporte técnico QSC "OnLine"

1200-14400 bps; 8N1
+(714) 668-7567
(800) 856-6003

CompuServe: GO QSCAUDIO

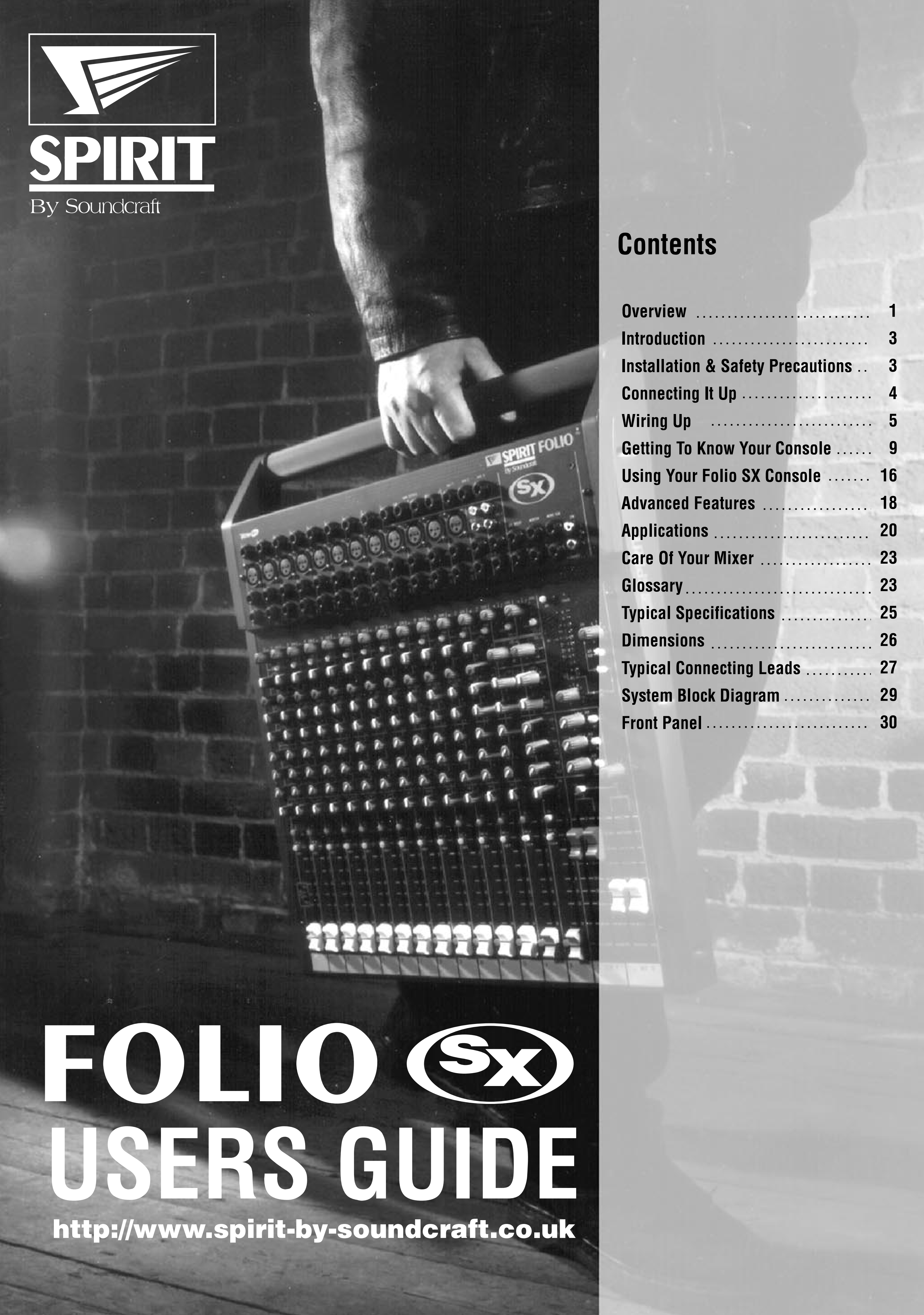
E-mail:

info@qscaudio.com
service@qscaudio.com
qsc_audio@compuserve.com

QSC Audio Products, Inc. 1675 MacArthur Boulevard Costa Mesa, California 92626 USA

"QSC" and the QSC logo are registered with the U.S. Patent and Trademark Office.
©1998 & 1999 QSC Audio Products, Inc.

SPIRIT
By Soundcraft

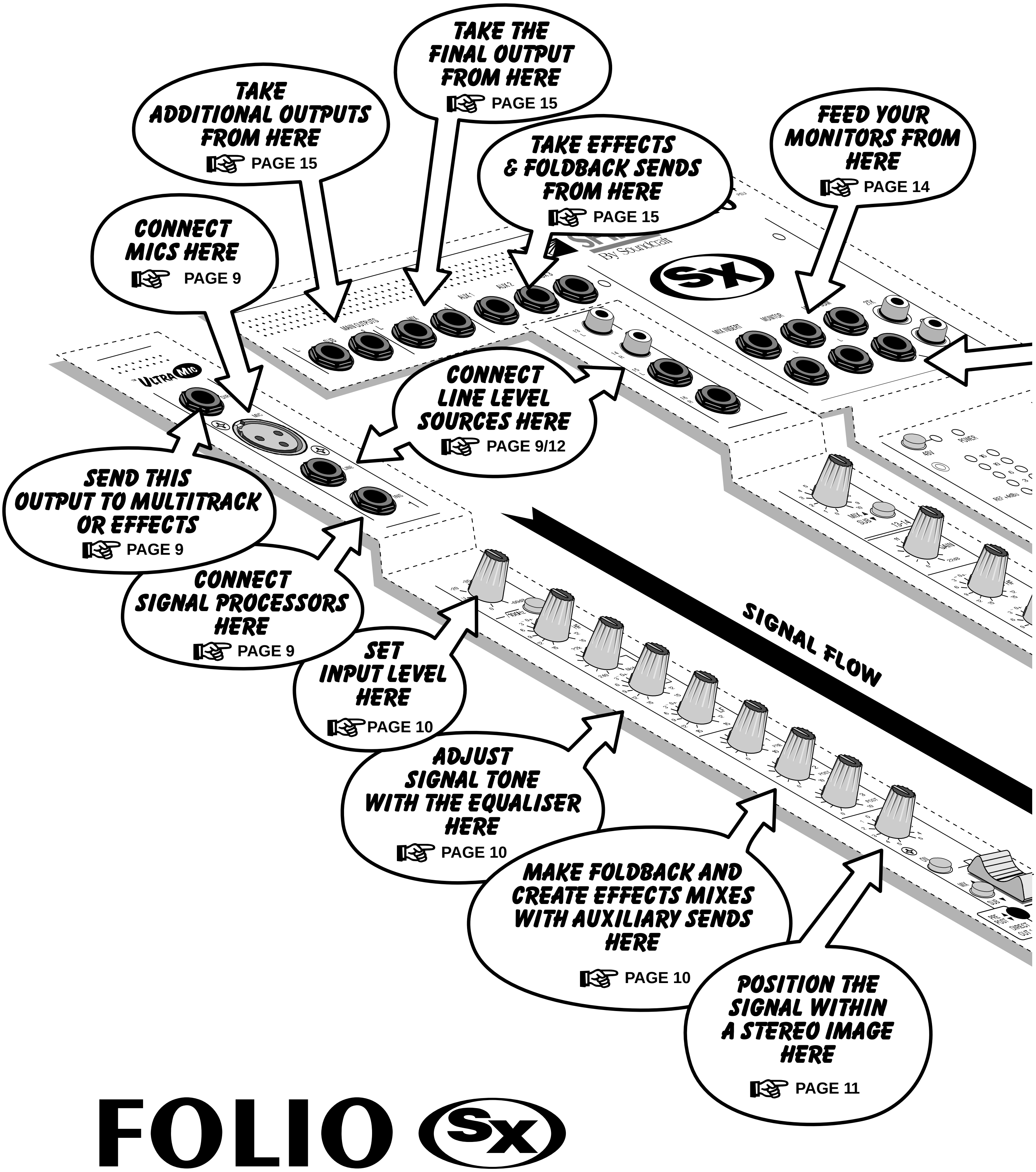


Contents

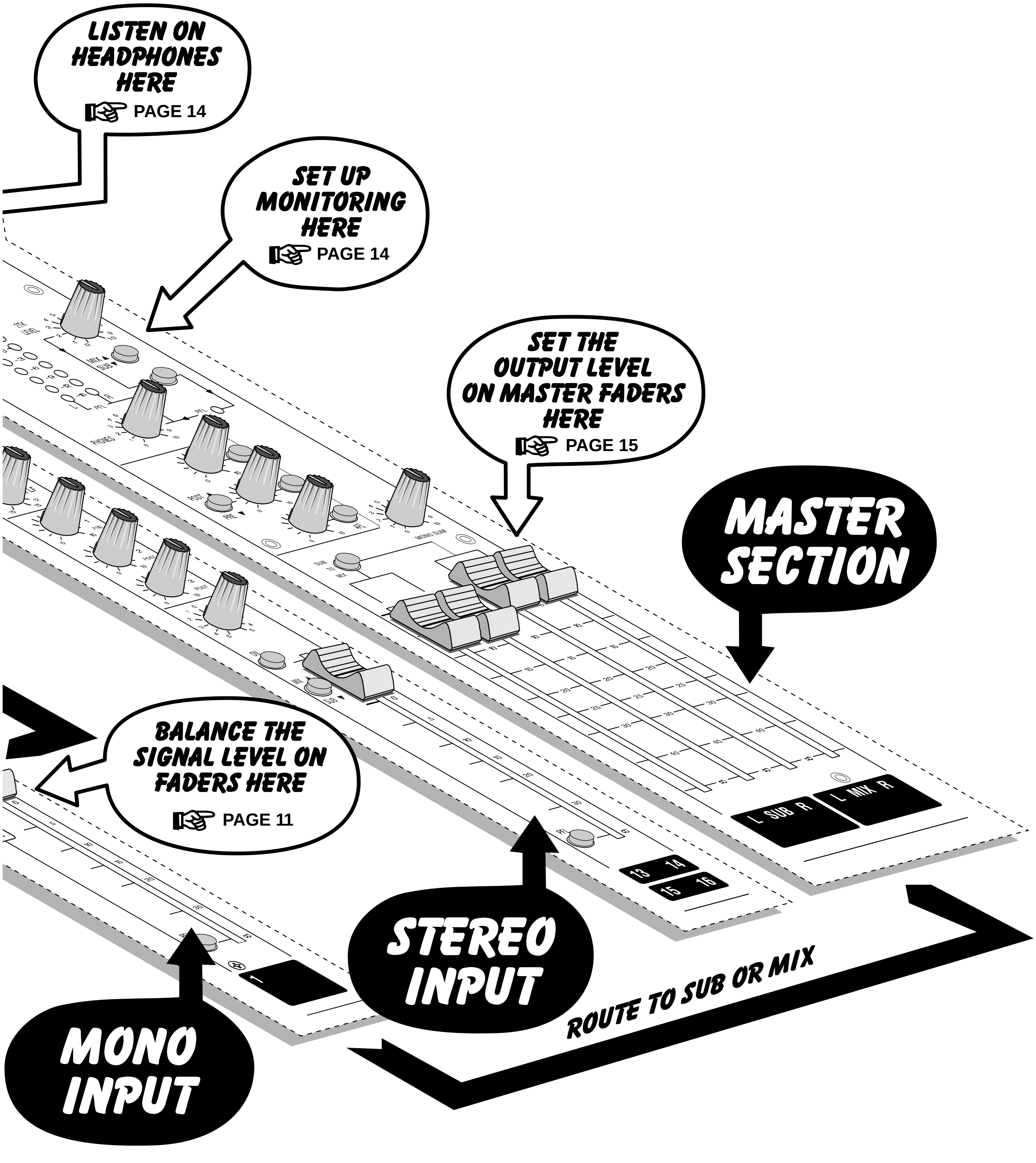
Overview 1
Introduction 3
Installation & Safety Precautions .. 3
Connecting It Up 4
Wiring Up 5
Getting To Know Your Console 9
Using Your Folio SX Console 16
Advanced Features 18
Applications 20
Care Of Your Mixer 23
Glossary 23
Typical Specifications 25
Dimensions 26
Typical Connecting Leads 27
System Block Diagram 29
Front Panel 30

FOLIO SX USERS GUIDE

http://www.spirit-by-soundcraft.co.uk


TAKE ADDITIONAL OUTPUTS FROM HERE ☞ PAGE 15
TAKE THE FINAL OUTPUT FROM HERE ☞ PAGE 15
TAKE EFFECTS & FOLDBACK SENDS FROM HERE ☞ PAGE 15
FEED YOUR MONITORS FROM HERE ☞ PAGE 14
CONNECT MICS HERE ☞ PAGE 9
CONNECT LINE LEVEL SOURCES HERE ☞ PAGE 9/12
SEND THIS OUTPUT TO MULTITRACK OR EFFECTS ☞ PAGE 9
CONNECT SIGNAL PROCESSORS HERE ☞ PAGE 9
SET INPUT LEVEL HERE ☞ PAGE 10
SIGNAL FLOW
ADJUST SIGNAL TONE WITH THE EQUALISER HERE ☞ PAGE 10
MAKE FOLDBACK AND CREATE EFFECTS MIXES WITH AUXILIARY SENDS HERE ☞ PAGE 10
POSITION THE SIGNAL WITHIN A STEREO IMAGE HERE ☞ PAGE 11
Sx
SPIRIT By Soundcraft
ULTRA MIC
MAIN OUTPUTS
AUX1
AUX2
MIX INSERT
MONITOR
POWER

FOLIO Sx OVERVIEW


LISTEN ON HEADPHONES HERE ☞ PAGE 14
SET UP MONITORING HERE ☞ PAGE 14
SET THE OUTPUT LEVEL ON MASTER FADERS HERE ☞ PAGE 15
MASTER SECTION
BALANCE THE SIGNAL LEVEL ON FADERS HERE ☞ PAGE 11
STEREO INPUT
MONO INPUT
ROUTE TO SUB OR MIX
13 14
15 16
L SUB R
L MIX R

INTRODUCTION

Thank you for purchasing a SPIRIT FOLIO SX mixer, brought to you with pride by the SPIRIT team of Andy, Colin, Chris, James, Simon, Mukesh, Graham, Martin, Paul, Matt and Peter, with the support of many others - we hope you will have as much fun using it!

Owning a SPIRIT console brings you the expertise and support of one of the industry's leading manufacturers and the results of over 23 years experience supporting some of the biggest names in the business.

Built to the highest standards using quality components and new surface mount technology, FOLIO SX is designed to be as easy to use as possible, but some time spent NOW, looking through this manual and getting to know your new mixer will give you lots of helpful tips and confidence, away from the pressures of a live session. Don't be afraid to experiment to find out how each control affects the sound - this will only extend your creativity and help you to get the best from your mixer.

INSTALLATION AND SAFETY PRECAUTIONS

INSTALLING THE MIXER

Correct connection and positioning of your mixer is important for successful and trouble-free operation. The following sections are intended to give guidance with cabling, connections and configuration of your mixer.

☐ Choose the mains supply for the sound system with care, and do not share sockets or earthing with lighting dimmers.

☐ Position the mixer where the sound can be heard clearly.

☐ Run audio cables separately from dimmer wiring, using balanced lines wherever possible. If necessary, cross audio and lighting cables at right angles to minimise the possibility of interference. Keep unbalanced cabling as short as possible.

☐ Check your cables regularly and label each end for easy identification.

SAFETY PRECAUTIONS

For your own safety and to avoid invalidation of the warranty please read this section carefully.

The FOLIO SX mixer must only be connected through the Power Supply supplied.

The wires in the mains lead are coloured in accordance with the following code:

Earth:	**Green and Yellow**	**(Green/Yellow - US)**
Neutral:	**Blue**	**(White - US)**
Live:	**Brown**	**(Black - US)**

As the colours of the wires in the mains lead may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

The wire which is coloured Green and Yellow must be connected to the terminal in the plug which is marked with the letter E or by the earth symbol.

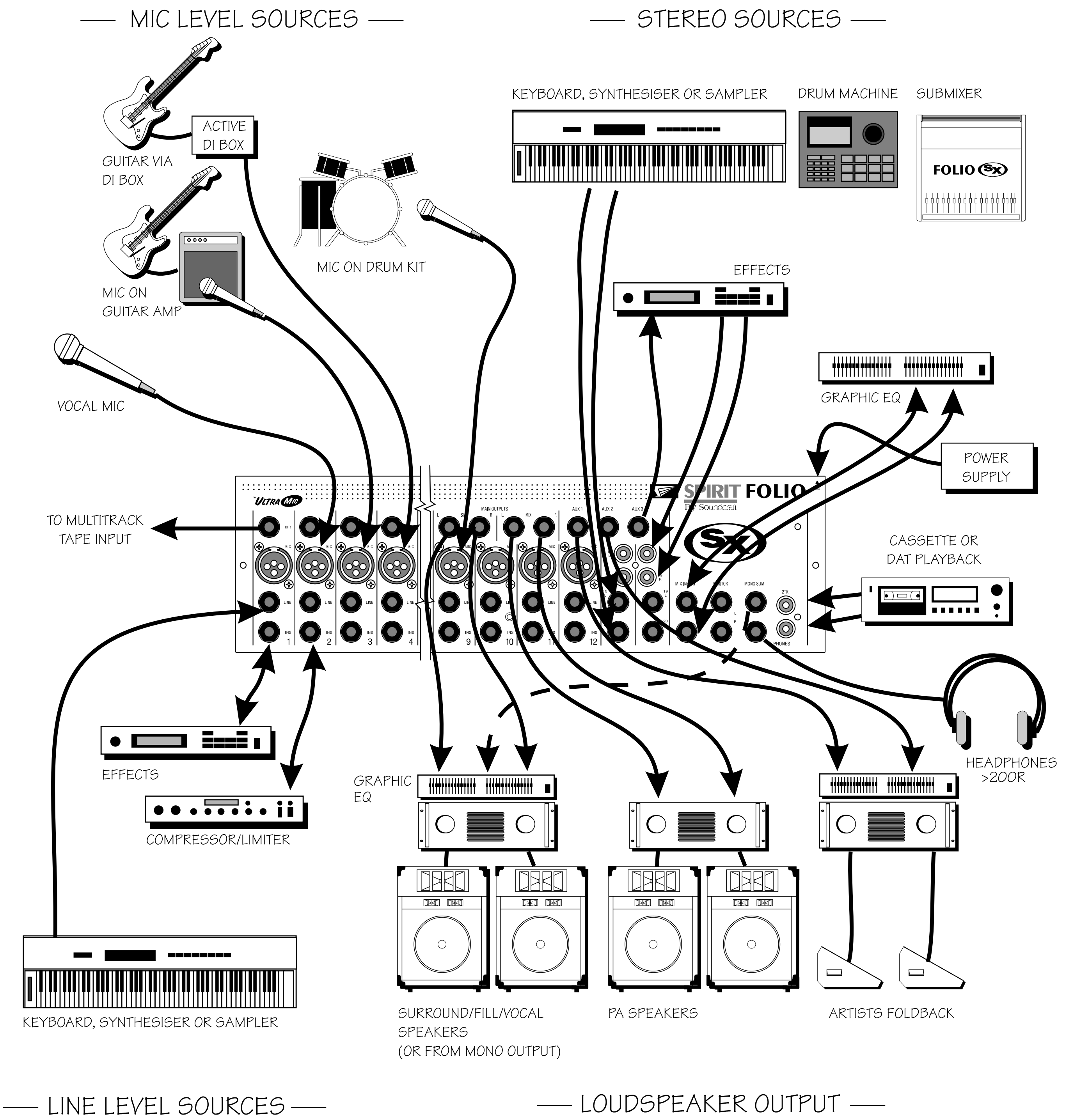
The wire which is coloured Blue must be connected to the terminal in the plug which is marked with the letter N.

The wire which is coloured Brown must be connected to the terminal in the plug which is marked with the letter L.

Ensure that these colour codings are followed carefully in the event of the plug being changed.

To avoid the risk of fire, replace the mains fuse only with the correct value fuse, as indicated on the power supply.

CONNECTING IT UP

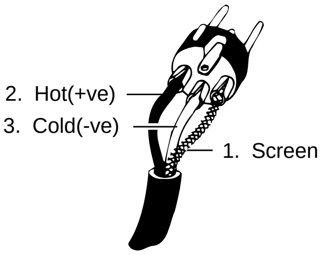

MIC LEVEL SOURCES
STEREO SOURCES
ACTIVE DI BOX
GUITAR VIA DI BOX
MIC ON GUITAR AMP
MIC ON DRUM KIT
VOCAL MIC
KEYBOARD, SYNTHESISER OR SAMPLER
DRUM MACHINE
SUBMIXER
FOLIO Sx
EFFECTS
GRAPHIC EQ
POWER SUPPLY
TO MULTITRACK TAPE INPUT
SPIRIT FOLIO
CASSETTE OR DAT PLAYBACK
EFFECTS
COMPRESSOR/LIMITER
GRAPHIC EQ
HEADPHONES >200R
KEYBOARD, SYNTHESISER OR SAMPLER
SURROUND/FILL/VOCAL SPEAKERS (OR FROM MONO OUTPUT)
PA SPEAKERS
ARTISTS FOLDBACK
LINE LEVEL SOURCES
LOUDSPEAKER OUTPUT

WIRING UP

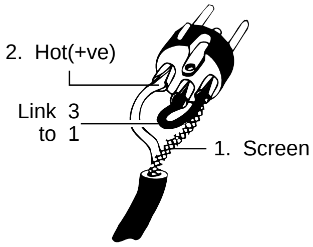
Please refer to pages 27/28 for additional wiring details.

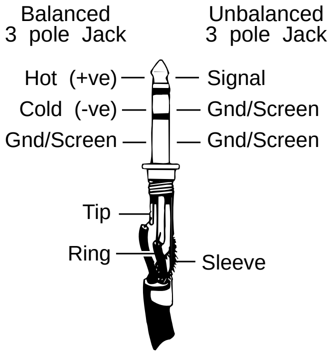
INPUTS

Balanced Mic
XLR

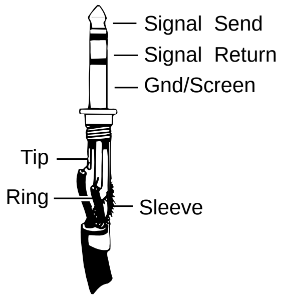


Unbalanced Mic
XLR





Inserts



MIC INPUT

The mic input accepts XLR-type connectors and is designed to suit a wide range of BALANCED or UNBALANCED low-level signals, whether from delicate vocals requiring the best low-noise performance or close-miked drum kits needing maximum headroom. Professional dynamic, condenser or ribbon mics are best because these will be LOW IMPEDANCE. While you can use low-cost HIGH IMPEDANCE mics, you do not get the same degree of immunity to interference on the microphone cable and as a result the level of background noise may be higher. If you turn the PHANTOM POWER on, the socket provides a suitable powering voltage for professional condenser mics.

DO NOT use UNBALANCED sources with the phantom power switched on.
The voltage on pins 2 & 3 of the XLR connector may cause serious damage.
BALANCED dynamic mics may normally be used with phantom power
switched on (contact your microphone manufacturer for guidance).

The input level is set using the input SENS knob.

The LINE input offers the same gain range as the MIC input, but at a higher input impedance. This is suitable for most line level sources, and provides the gain needed for lower level keyboards and high impedance microphones.

WARNING - Start with the input SENS knob turned fully anticlockwise when
plugging high level sources into the LINE input to avoid overloading the
input channel or giving you a very loud surprise!

LINE INPUT

Accepts 3-pole `A' gauge jacks, or 2-pole mono jacks which will automatically ground the 'cold' input. Use this input for sources other than mics, such as keyboards, drum machines, synths, tape machines or guitars. The input is BALANCED for low noise and immunity from interference, but you can use UNBALANCED sources by wiring up the jacks as shown, although you should then keep cable lengths as short as possible to minimise interference pick-up on the cable. Note that the ring must be grounded if the source is unbalanced. Set the input level using the SENS knob, starting with the knob turned fully anticlockwise. Unplug any MIC connection when using the LINE input.

INSERT POINT

The unbalanced, pre-EQ insert point is a break in the channel signal path, allowing limiters, compressors, special EQ or other signal processing units to be added in the signal path. The Insert is a 3-pole 'A' gauge jack socket which is normally bypassed. When a jack is inserted, the signal path is broken, just before the EQ section.

The signal from the channel appears on the TIP of the plug and is returned on the RING, with the sleeve as a common ground.

The Send may be tapped off as an alternative pre-fade, pre-EQ direct output if required, using a lead with tip and ring shorted together so that the signal path is not interrupted.

STEREO INPUTS 13/14 & 17/18

These accept RCA phono jacks to allow easy connection to hi-fi equipment or DAT players. The input is unbalanced, and ideal for pre-show music sources or signals that do not require any EQ or effects.

STEREO INPUTS 15/16 & 19/20

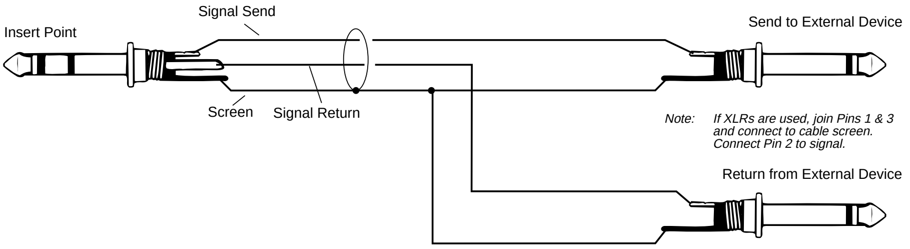
These accept 3-pole `A' gauge jacks, or 2-pole mono jacks which will automatically ground the 'cold' input. Use these inputs for sources such as keyboards, drum machines, synths, tape machines or as returns from processing units. The input is BALANCED for low noise and immunity from interference, but you can use UNBALANCED sources by wiring up the jacks as shown, although you should then keep cable lengths as short as possible to minimise interference pick-up on the cable. Note that the ring must be grounded if the source is unbalanced.

Mono sources can be fed to both paths by plugging into the Left jack only.

MIX INSERTS

The unbalanced, pre-fade Mix insert point is a break in the output signal path to allow the connection of, for example, a compressor/limiter or graphic equaliser. The Insert is a 3-pole 'A' gauge jack socket which is normally bypassed. When a jack is inserted, the signal path is broken, just before the mix fader.

The mix signal appears on the TIP of the plug and is returned on the RING. A 'Y' lead may be required to connect to equipment with separate send and return jacks as shown below:

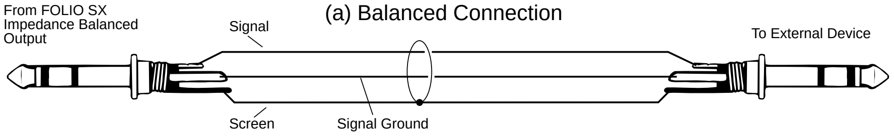

Insert Point
Signal Send
Screen
Signal Return
Send to External Device
Return from External Device

Note: *If XLRs are used, join Pins 1 & 3 and connect to cable screen. Connect Pin 2 to signal.*

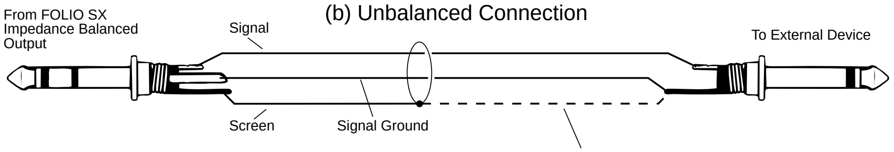
MIX & SUB OUTPUTS

The MIX and SUB outputs are on 3-pole 3-pole 'A' gauge jack sockets, wired as shown, and incorporate impedance balancing, allowing long cable runs to balanced amplifiers and other equipment.

(a) Balanced Connection


From FOLIO SX
Impedance Balanced
Output
Signal
To External Device
Screen
Signal Ground

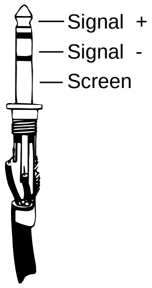
(b) Unbalanced Connection


From FOLIO SX
Impedance Balanced
Output
Signal
To External Device
Screen
Signal Ground

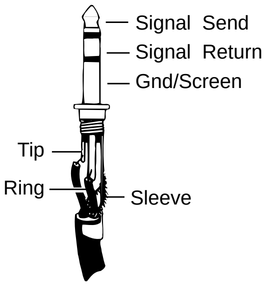
Experience has shown that sometimes it is better not to connect screen at external device end.

OUTPUTS

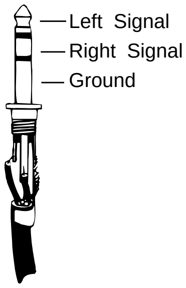
Mix & Sub Outputs
Aux Outputs
Mono Output


Signal +
Signal -
Screen

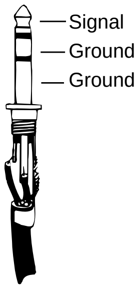
Mix Inserts


Signal Send
Signal Return
Gnd/Screen
Tip
Ring
Sleeve

Headphones

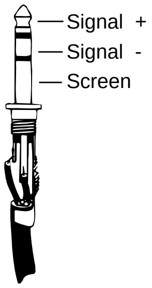

Left Signal
Right Signal
Ground

Monitor Outputs
Direct Outputs

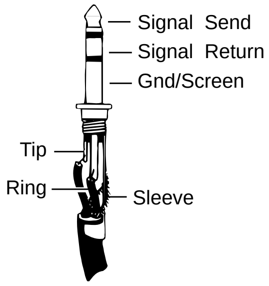

Signal
Ground
Ground

OUTPUTS

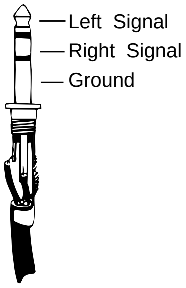
Mix & Sub Outputs
Aux Outputs
Mono Output


Signal +
Signal -
Screen

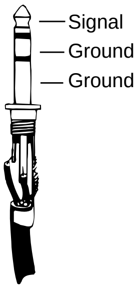
Mix Inserts


Signal Send
Signal Return
Gnd/Screen
Tip
Ring
Sleeve

Headphones


Left Signal
Right Signal
Ground

Monitor Outputs
Direct Outputs


Signal
Ground
Ground

AUX OUTPUTS

The Aux outputs are on 3-pole ‘A’ gauge jack sockets, wired as shown on the left, and incorporate impedance balancing, allowing long cable runs to balanced amplifiers and other equipment.

CHANNEL DIRECT OUTPUTS

The Direct outputs are on 3-pole ‘A’ gauge jack sockets, wired as shown on the left, and are unbalanced.

HEADPHONES

The PHONES output is a 3-pole ‘A’ gauge jack, wired as a stereo output as shown, suitable for headphones of 200Ω or greater. 8Ω headphones are not recommended.

POLARITY (PHASE)

You will probably be familiar with the concept of polarity in electrical signals and this is of particular importance to balanced audio signals. Just as a balanced signal is highly effective at cancelling out unwanted interference, so two microphones picking up the same signal can cancel out, or cause serious degradation of the signal if one of the cables has the +ve and -ve wires reversed. This phase reversal can be a real problem when microphones are close together and you should therefore take care always to connect pins correctly when wiring audio cables.

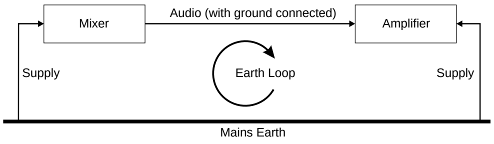
GROUNDING AND SHIELDING

For optimum performance use balanced connections where possible and ensure that all signals are referenced to a solid, noise-free earthing point and that all signal cables have their screens connected to ground. In some unusual circumstances, to avoid earth `loops’ ensure that all cable screens and other signal earths are connected to ground only at their source and not at both ends.

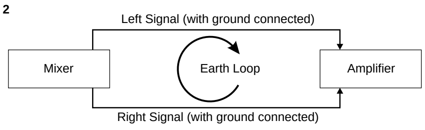
If the use of unbalanced connections is unavoidable, you can mimimise noise by following these wiring guidelines:

- On INPUTS, unbalance at the source and use a twin, screened cable as though it were balanced.
- On OUTPUTS, connect the signal to the +ve output pin, and the ground of the output device to -ve. If a twin screened cable is used, connect the screen only at the mixer end.
- Avoid running audio cables or placing audio equipment, close to thyristor dimmer units or power cables.
- Noise immunity is improved significantly by the use of low impedance sources, such as good quality professional microphones or the outputs from most modern audio equipment. Avoid cheaper high impedance microphones, which may suffer from interference over long cable runs, even with well-made cables.
- Grounding and shielding is still seen as a black art, and the suggestions above are only guidelines. If your system still hums, an earth loop is the most likely cause. Two examples of how an earth loop can occur are shown on the right

Example 1


Mixer
Audio (with ground connected)
Amplifier
Supply
Earth Loop
Supply
Mains Earth

Example 2


Left Signal (with ground connected)
Mixer
Earth Loop
Amplifier
Right Signal (with ground connected)

FAULT FINDING GUIDE

Repairing a sound mixing console requires specialist skills, but basic Fault Finding is within the scope of any user if a few basic rules are followed.

- Get to know the Block Diagram of your console (see page 29).
- Get to know what each component in the system is supposed to do.
- Learn where to look for common trouble spots.

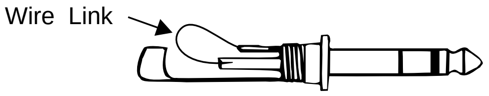
The Block Diagram is a representative sketch of all the components of the console, showing how they connect together and how the signal flows through the system. Once you have become familiar with the various component blocks you will find the Block Diagram quite easy to follow and you will have gained a valuable understanding of the internal structure of the console.

Each Component has a specific function and only by getting to know what each part is supposed to do will you be able to tell if there is a genuine fault! Many `faults' are the result of incorrect connection or control settings which may have been overlooked.

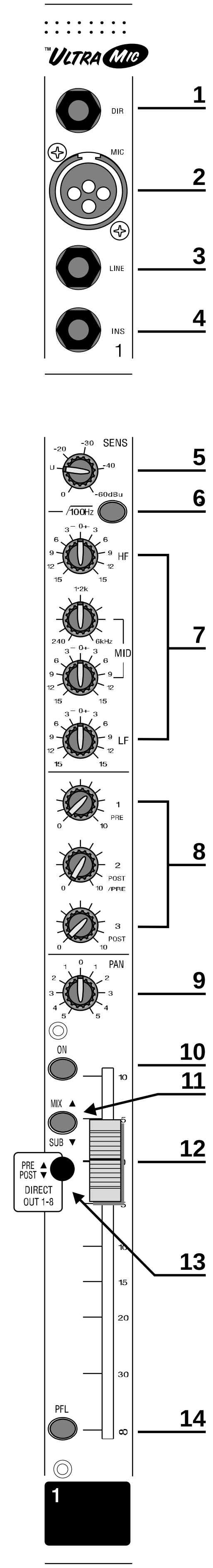
Basic Troubleshooting is a process of applying logical thought to the signal path through the console and tracking down the problem by elimination.

- Swap input connections to check that the source is really present. Check both Mic and Line inputs.
- Eliminate sections of the channel by using the insert point to re-route the signal to other inputs that are known to be working.
- Route channels to different outputs or to auxiliary sends to identify problems on the Master section.
- Compare a suspect channel with an adjacent channel which has been set up identically. Use PFL and AFL to monitor the signal in each section.

Insert contact problems may be checked by using a dummy jack with tip and ring shorted together as shown below. If the signal appears when the jack is inserted it shows that there is a problem with the normalling contacts on the jack socket, caused by wear or damage, or often just dirt or dust. Keep a few in your gig tool box.



Dummy Insert Bypass Jack


ULTRA MIC
1
DIR
2
MIC
3
LINE
4
INS
1
SENS
-20
-30
-40
0
-60dBu
5
6
100Hz
HF
1·2k
240
6kHz
MID
LF
7
1
PRE
2
POST /PRE
3
POST
8
PAN
9
ON
10
MIX ▲
SUB ▼
11
PRE ▲
POST ▼
DIRECT OUT 1-8
12
13
15
20
30
PFL
∞
14
1

GETTING TO KNOW YOUR CONSOLE

MONO INPUT CHANNEL

1 DIRECT OUTPUT

The first eight channels have a dedicated Direct Output which allows direct connection to external devices, for example to feed Tape Machines or effects units. See the 'Advanced Features' section on page 18 for a full explanation of these outputs.

2 MIC INPUT

The mic input accepts XLR-type connectors and is designed to suit a wide range of BALANCED or UNBALANCED signals. Professional dynamic, condenser or ribbon mics are best because these will be LOW IMPEDANCE. You can use low-cost HIGH IMPEDANCE mics, but the level of background noise will be higher. If you turn the PHANTOM POWER on (top right-hand side of the mixer) the socket provides a suitable powering voltage for professional condenser mics.

ONLY connect condenser microphones with the +48V powering OFF, and ONLY turn the +48V powering on or off with all output faders DOWN, to prevent damage to the mixer or external devices.

TAKE CARE when using unbalanced sources, which may be damaged by the phantom power voltage on pins 2 & 3 of the XLR connector.

Unplug any mics if you want to use the LINE Input. The input level is set using the SENS knob.

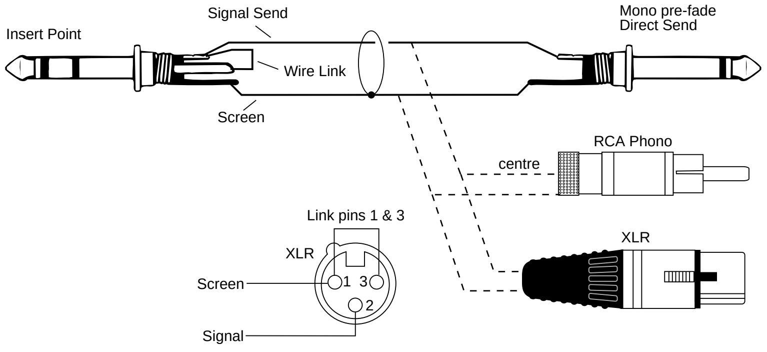
3 LINE INPUT

Accepts 3-pole `A' gauge (TRS) jacks. Use this input for sources other than mics, such as keyboards, drum machines, synths, tape machines or guitars. The input is BALANCED for low noise and top quality from professional equipment, but you can use UNBALANCED sources by wiring up the jacks as shown below, although you should then keep cable lengths as short as possible. Unplug anything in the MIC input if you want to use this socket. Set the input level using the SENS knob.

4 INSERT POINT (ALTERNATIVE DIRECT SEND)

The unbalanced, pre-EQ insert point is a break in the channel signal path, allowing limiters, compressors, special EQ or other signal processing units to be added in the signal path. The Insert is a 3-pole 'A' gauge jack socket which is normally bypassed. When a jack is inserted, the signal path is broken, just before the EQ section.

The Send may be tapped off as an alternative pre-fade, pre-EQ direct output if required, using a lead with tip and ring shorted together so that the signal path is not interrupted (see below).


Insert Point
Signal Send
Wire Link
Screen
Mono pre-fade Direct Send
RCA Phono
centre
XLR
Link pins 1 & 3
XLR
Screen
1
3
2
Signal

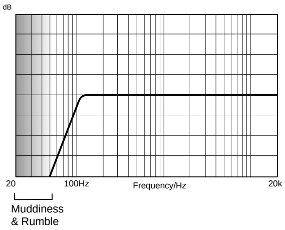
5 SENS

This knob sets how much of the source signal is sent to the rest of the mixer. Too high, and the signal will distort as it overloads the channel. Too low, and the level of any background hiss will be more noticeable and you may not be able to get enough signal level to the output of the mixer.

Setting the knob to the `U' mark gives unity gain for the LINE input. Note that some sound equipment, particularly that intended for domestic use, operates at a lower level (-10dBV) than professional equipment and will therefore need a higher gain setting to give the same output level.

See `Initial Set Up' on page 16 to learn how to set SENS correctly.

6 100Hz HI-PASS FILTER

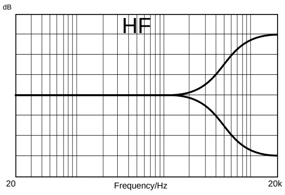


Pressing this switch activates a steep 18dB per octave filter which reduces the level of bass frequencies only, and is a real bonus for a such a small mixer. Use this in live PA situations to clean up the mix, reducing stage rumble or `popping' from microphones.

7 EQUALISER

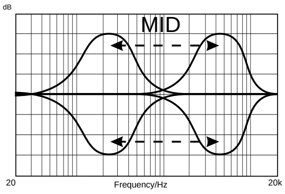
The Equaliser (EQ) allows fine manipulation of the sound, particularly to improve the sound in live PA applications where the original signal is often far from ideal and where slight boosting or cutting of particular voice frequencies can really make a difference to clarity. There are three sections giving the sort of control usually only found on much larger mixers. The EQ knobs can have a dramatic effect, so use them sparingly and listen carefully as you change any settings so that you get to know how they affect the sound.

HF EQ



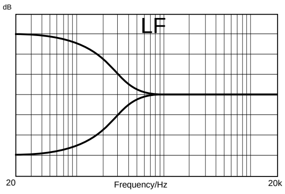
Turn to the right to boost high (treble) frequencies above 12kHz by up to 15dB, adding crispness to cymbals, vocals and electronic instruments. Turn to the left to cut by up to 15dB, reducing hiss or excessive sibilance which can occur with certain types of microphone. Set the knob in the centre-detented position when not required.

MID EQ



There are two knobs which work together to form a SWEPT MID EQ. The lower knob provides 15dB of boost and cut, just like the HF EQ knob, but the frequency at which this occurs can be set by the upper knob over a range of 240Hz to 6kHz. This allows some truly creative improvement of the signal in live situations, because this mid band covers the range of most vocals. Listen carefully as you use these controls together to find how particular characteristics of a vocal signal can be enhanced or reduced. Set the lower knob to the centre-detented position when not required.

LF EQ



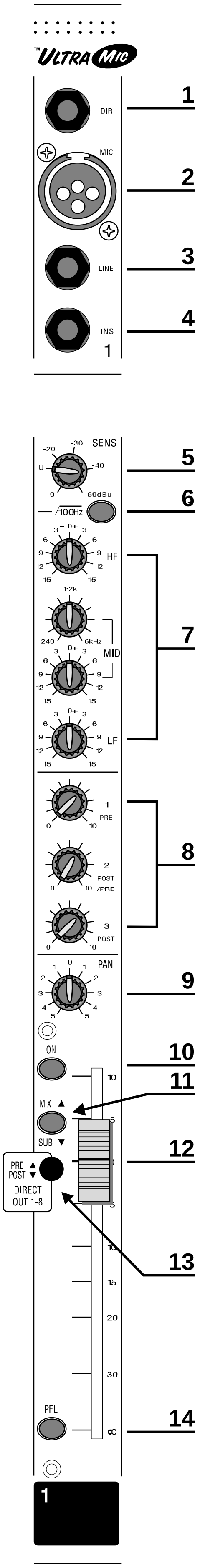
Turn to the right to boost low (bass) frequencies below 60Hz by up to 15dB, adding warmth to vocals or extra punch to synths, guitars and drums. Turn to the left to cut low frequencies by up to 15dB for reducing hum, stage rumble or to improve a mushy sound. Set the knob to the centre-detented position when not required.

8 AUX SENDS

These are used to set up a separate mixes for FOLDBACK, EFFECTS or recording, and the combination of each the Aux Send is mixed to the respective Aux Output at the rear of the mixer. For Effects it is useful for the signal to fade up and down with the fader (this is called POST-FADE), but for Foldback or Monitor feeds it is important for the send to be independent of the fader (this is called PRE-FADE).

AUX SEND 1

This is always PRE-FADE and therefore most appropriate for foldback or monitor mixes or external submix.



AUX SEND 2

This is normally POST-FADE, and would typically be used as an effects send, but may be switched globally to PRE-FADE using the POST/PRE switch on the Master section, providing a second foldback or monitor send if required.

AUX SEND 3

This is always POST-FADE for effects sends, external submix (or for Centre Voice cluster or mono Tape mix).

9 PAN

This control sets the amount of the channel signal feeding the Right and Left MIX or SUB buses, allowing you to move the source smoothly across the stereo image. When the control is turned fully right or left you are able to route the signal at unity gain to either left or right outputs individually.

10 ON

All outputs from the channel except inserts are muted when the ON switch is released and enabled when the switch is down, allowing levels to be pre-set before the before the signal is required.

11 MIX / SUB

The channel signal may be routed to either the main MIX (switch UP) or SUB buses (switch DOWN). The left/right balance is set by the PAN control (see above) allowing the signal to be sent proportionately to any of the buses.

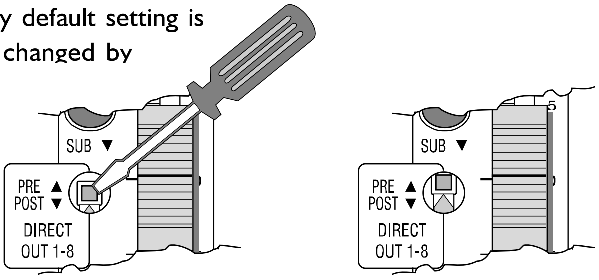
12 FADER

The 100mm FADER, with a custom-designed law to give even smoother control of the overall signal level in the channel strip, allows precise balancing of the various source signals being mixed to the Master Section. You get most control when the input SENSITIVITY is set up correctly, giving full travel on the fader. See the `Initial Set Up' section on page 16 for help in setting a suitable signal level.

13 DIRECT PRE/POST

See Direct Output (1) and 'Advanced Features' on page 18 for an explanation of the Direct Output. The output may be selected as either PRE or POST fader. The factory default setting is POST-FADER, but this may be changed by moving an internal switch as shown on the right.

Remove the protective grommet to gain access to the switch and move the switch carefully upwards using a small screwdriver or the end of a pencil. It is advisable to turn the unit off before making this adjustment.

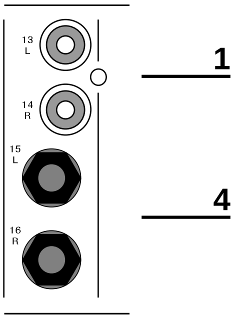


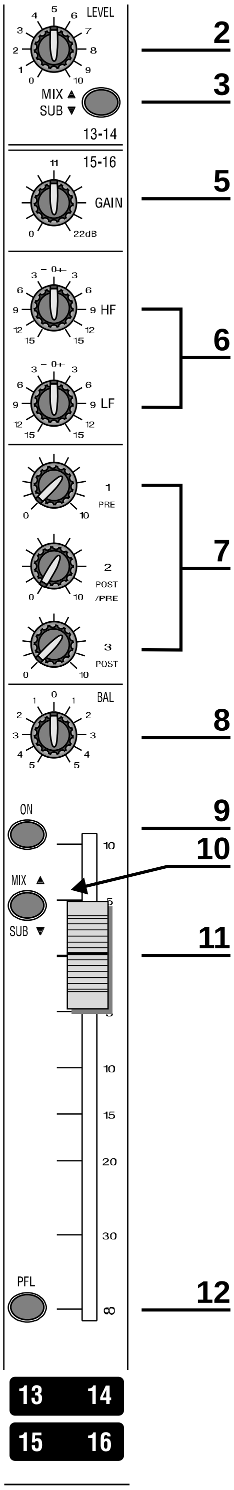
POST-FADE PRE-FADE

Always check all outputs after adjusting to ensure that switching has been successful.

14 PFL

When the latching PFL switch is pressed, the pre-fade signal is fed to the headphones, control room output and meters, where it replaces the selected monitor source (MIX, SUB or 2TK). The PFL/AFL LED on the Master section illuminates to warn that a PFL is active. This is a useful way of listening to any required input signal without interrupting the main mix, for making adjustments or tracing problems.





STEREO INPUT CHANNEL

Each stereo input channel comprises two pairs of inputs per channel strip:

1 INPUTS 13/14 & 17/18

These inputs are unbalanced on RCA phono connectors, and are intended for use with CD players, DAT machines or Hi-Fi equipment. Alternatively they may be used as simple effects returns or stereo instrument inputs.

2 LEVEL

The LEVEL control sets the level of the channel signal.

3 MIX/SUB

The MIX/SUB switch allows the signal to be routed to either the main MIX (switch up) or SUB mix (switch down).

4 INPUTS 15/16 & 19/20

These inputs accept 3-pole `A' gauge (TRS) jacks. Use these inputs for sources such as keyboards, drum machines, synths, tape machines or processing units. The inputs are BALANCED for low noise and top quality from professional equipment, but you can use UNBALANCED sources by wiring up the jacks as shown in the "Wiring it Up" section earlier in this manual, although you should then keep cable lengths as short as possible. Mono sources may be used by plugging into the left jack only.

5 GAIN

The GAIN control sets the input level to the channel, allowing matching to a wide range of line level sources.

6 EQUALISER

HF EQ

Turn to the right to boost high (treble) frequencies, adding crispness to percussion from drum machines, synths and electronic instruments. Turn to the left to cut these frequencies, reducing hiss or excessive brilliance. Set the knob in the centre-detented position when not required. The control has a shelving response giving 15dB of boost or cut at 12kHz.

LF EQ

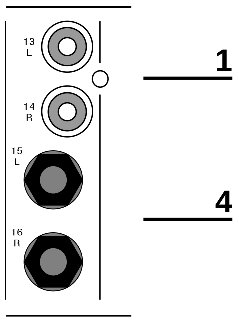
Turn to the right to boost low (bass) frequencies, adding extra punch to synths, guitars and drums. Turn to the left to reduce hum, boominess or improve a mushy sound. Set the knob to the centre-detented position when not required. The control has a shelving response giving 15dB of boost or cut at 60Hz.

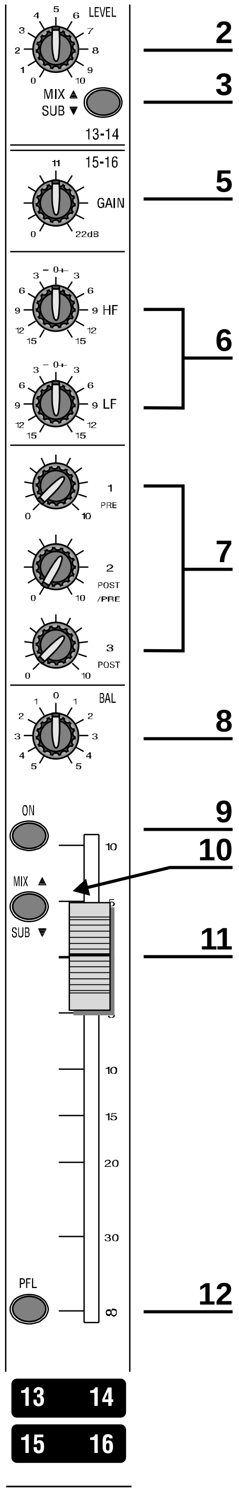
7 AUX SENDS

These are used to set up a separate mixes for FOLDBACK, EFFECTS or recording, and the combination of each the Aux Send is mixed to the respective Aux Output at the rear of the mixer. For Effects it is useful for the signal to fade up and down with the fader (this is called POST-FADE), but for Foldback or Monitor feeds it is important for the send to be independent of the fader (this is called PRE-FADE).

AUX SEND 1

This is always PRE-FADE and therefore most appropriate for foldback or monitor mixes.





AUX SEND 2

This is normally POST-FADE, and would typically be used as an effects send, but may be switched globally to PRE-FADE using the POST/PRE switch on the Master section, providing a second foldback or monitor send if required.

AUX SEND 3

This is always POST-FADE.

8 BALANCE

This control sets the amount of the channel signal feeding the Right and Left MIX or SUB buses, allowing you to balance the source in the stereo image. When the control is turned fully right or left you feed only that side of the signal to the mix. Unity gain is provided by the control in the centre-detented position.

9 ON

All outputs from the channel except inserts are muted when the ON switch is released and enabled when the switch is down.

10 MIX / SUB

The channel signal may be routed in stereo to either the main MIX (switch UP) or SUB buses (switch DOWN). The left/right balance is set by the BAL control (see above).

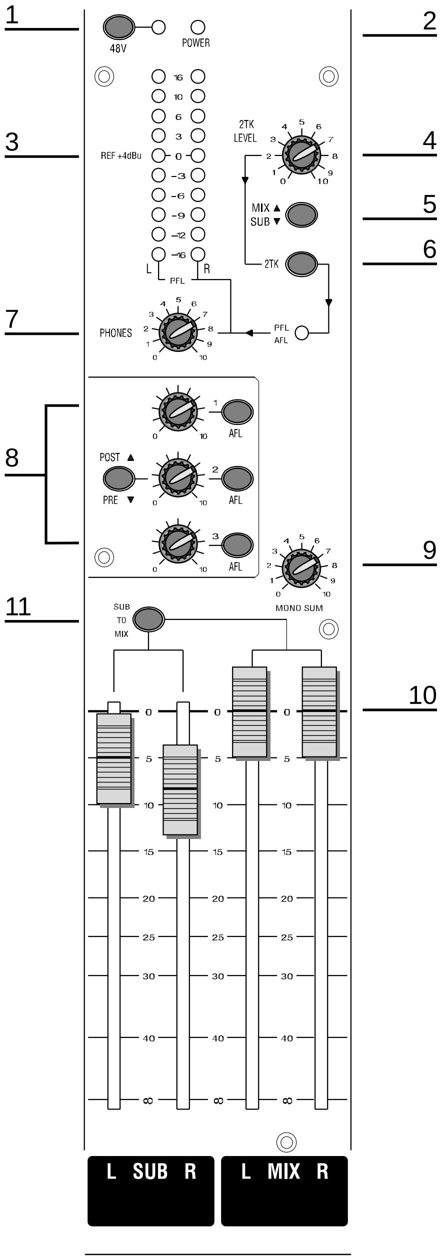
11 FADER

The 100mm FADER gives you smooth control of the overall signal level in the channel strip, allowing precise balancing of the various source signals being mixed to the Master Section. It is important that the input level is set correctly to give maximum travel on the fader which should normally be used at around the `0' mark. See the `Initial Set Up' section on page 16 for help in setting the right level.

12 PFL

When the latching PFL switch is pressed, the pre-fade signal is fed in mono to the headphones, control room output and meters, where it replaces the selected monitor source (MIX, SUB or 2TK). The PFL/AFL LED on the Master section illuminates to warn that a PFL is active. The Left and Right meters display the PFL signal in mono. This is a useful way of listening to any required input signal without interrupting the main mix, for making adjustments or tracing problems.

MASTER SECTION



1 PHANTOM POWER

Many professional condenser mics need PHANTOM POWER, which is a method of sending a powering voltage down the same wires as the mic signal. Press the switch to enable the +48V power to all of the MIC inputs. The adjacent LED illuminates when the power is active. TAKE CARE when using unbalanced mics which may be damaged by the phantom power voltage. Balanced dynamic mics can normally be used with phantom power switched on (contact your microphone manufacturer for guidance)

Mics should always be plugged in, and all output faders set to minimum before switching the Phantom Power ON to avoid damage to external equipment

2 POWER INDICATOR

This LED lights to show when power is connected to the console.

3 BARGRAPH METERS

The three colour peak reading BARGRAPH METERS normally follow the Monitor selection to show the level of the MIX RIGHT and MIX LEFT outputs, giving you a constant warning of excessive peaks in the signal which might cause overloading. Aim to keep the signal just touching the amber segments at peak levels for best performance.

Similarly, if the output level is too low and hardly registering at all on the meters, the level of background noise may become significant. Take care to set up the input levels for best performance.

When any PFL switch is pressed, the meters switch to show the selected PFL signal on both meters, in mono.

4 2TK LEVEL

The rotary control sets the level of the 2 Track Tape input, which is routed to the headphones, monitor outputs and meters. These inputs, on RCA phono connectors are an ideal to connect the playback of a tape machine for monitoring.

5 MIX/SUB

When this switch is released the meters and headphones monitor the Mix signal. Alternatively, pressing the switch selects the SUB mix as the monitor source.

6 2TK

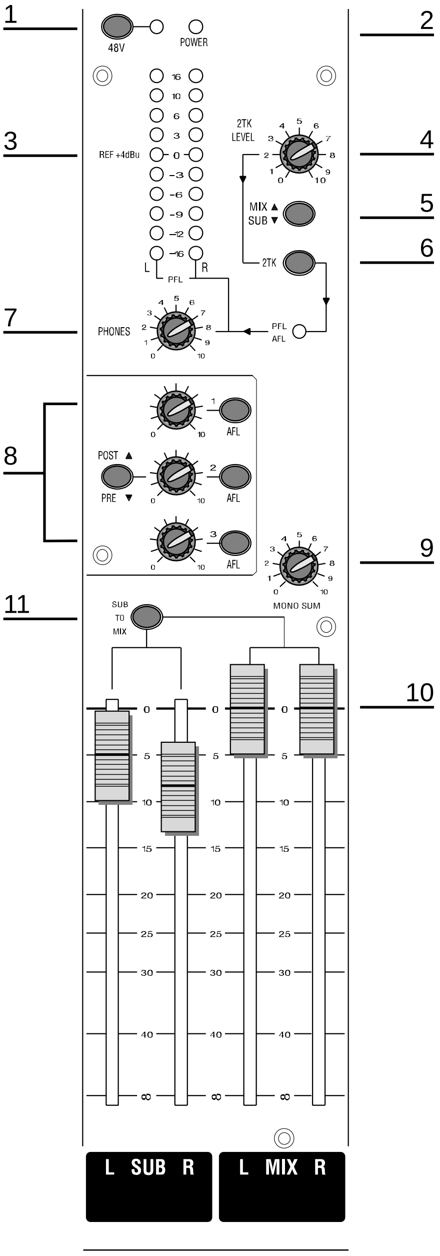
Pressing the 2TK switch selects the 2TK input as the monitor source, instead of the MIX or SUB signal (see MIX/SUB above).

7 MONITOR & PHONES LEVEL

This control sets the output level to the MONITOR LEFT & RIGHT outputs. If HEADPHONES are plugged into the PHNS jack the Monitor outputs are cut off, and the knob then sets the headphone listening level. When the PHONES are unplugged the Monitor output is restored.

8 AUX MASTERS

Each Aux output has a master output level control and associated AFL switch.



AUX AFLs

Just like the PFL switches on the channels, you can monitor each AUX output by pressing the AFL switch. This routes the AUX output signal to the MONITOR or PHONES, replacing any existing signal (normally the Monitor receives either MIX, SUB or 2TK, see above). The METERS also switch from the selected source to display the PFL/AFL signal and the PFL/AFL LED lights to warn that a PFL or AFL switch is pressed. When you release the switch the Monitor swaps back to the previous source.

AUX 2 POST/PRE

The input channels provide both Pre- and Post-Fade AUX 2 sends which may be selected desk-wide on the Master Section. Press the POST/PRE switch to make all of the AUX 2 Sends on the channel strips PRE-FADE. This means that they will all be unaffected by the position of the channel faders, making them ideal for FOLDBACK or MONITORING.

When the switch is released the AUX 2 Sends are all POST-FADE, and will fade up and down with the channel faders. This is more suitable for effects sends which need to fade out with the associated source.

9 MONO SUM

The Mix Left and Right signals are summed to a MONO output on a 3 pole ‘A’ gauge jack. Output level is set by the dedicated rotary control. Monitoring of the Mono output, if required, but be done at the external equipment it feeds, or the signal brought back to a spare console input.

10 MASTER FADERS

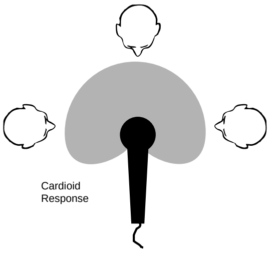
The MASTER FADERS set the final level of the MIX and SUB outputs, and separate faders are provided for each output. These should normally be set close to the `0’ mark if the input GAIN settings have been correctly set, to give maximum travel on the faders for smoothest control.

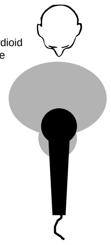
11 SUB TO MIX

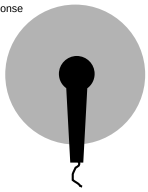
The SUB outputs may be used as independent outputs from the console, or used as true subgroups, routed to the main Mix outputs. When the switch is released the SUB signal feeds the SUB outputs only. With the switch pressed the post-fade Sub Left and Right signals are routed to Mix Left and Right respectively. The Sub outputs remain unaffected.

USING YOUR FOLIO SX CONSOLE

The final sound from your P.A. system can only ever be as good as the weakest link in the chain, and especially important is the quality of the source signal because this is the starting point of the chain. Just as you need to become familiar with the control functions of your mixer, so you must recognise the importance of correct choice of inputs, microphone placement and input channel settings. However, no amount of careful setting up can take account of the spontaneity and unpredictability of live performance and the mixer must be set up to provide `spare' control range to compensate for changing microphone position and the absorption effect of a large audience (different acoustic characteristics from sound check to show).


Cardioid Response


Hypercardioid Response


Omnidirectional Response

MICROPHONE PLACEMENT

Careful microphone placement and the choice of a suitable type of microphone for the job is one of the essentials of successful sound reinforcement. The diagrams on the left show the different pick-up patterns for the most common types of microphone. Cardioid microphones are sensitive to sound coming from in front, and hypercardioid microphones offer even greater directivity, with a small amount of pickup behind the microphone. These types are ideal for recording vocalists or instruments, where rejection of unwanted sounds and elimination of feedback is important. The aim should be to place the microphone as close as physically possible to the source, to cut out unwanted surrounding sounds, allow a lower gain setting on the mixer and avoid feedback. Also a well-chosen and well-placed microphone should not need any appreciable equalisation.

There are no exact rules - let your ears be the judge. In the end, the position that gives the desired effect is the correct position!

INITIAL SET UP

Once you have connected up your system (see the sections on connection and wiring earlier in this manual for guidance) you are ready to set initial positions for the controls on your mixer.

The front panel drawing inside the rear cover shows typical initial control positions which may be found a useful guide to setting up the mixer for the first time.

Set up individual input channel as follows:

- Connect your sources (microphone, keyboard etc.) to the required inputs.
  Note: Phantom powered mics should be connected before the +48V is switched on.
  Route the channel to Mix.
- Set Master faders at 0, input faders at 0, and set power amplifier levels to about 70%.
- Provide a typical performance level signal and press the PFL button on the first channel, monitoring the level on the bargraph meters.
- Adjust the input gain until the meter display is in the amber section, with occasional peaks to the first red LED at a typical maximum source level. This allows sufficient headroom to accommodate peaks and establishes the maximum level for normal operation (but see note below).

- Repeat this procedure on other channels as required. As more channels are added to the mix, the meters may move into the red section. Adjust the overall level using the Master Faders if necessary.
- Listen carefully for the characteristic sound of `feedback'. If you cannot achieve satisfactory input level setting without feedback, check microphone and speaker placement and repeat the exercise. If feedback persists, it may be necessary to use a Graphic Equaliser to reduce the system response at particular resonant frequencies.

Note: The initial settings should only be regarded as a starting point for your mix. It is important to remember that many factors affect the sound during a live performance, for instance the size of the audience!

You are now ready to start building the mix and this should be done progressively, listening carefully for each component in the mix and watching the meters for any hint of overload. If this occurs, back off the appropriate Channel Fader slightly until the level is out of the red segments, or adjust the Master Faders.

Remember that the mixer is a mixer, not an amplifier. Increasing the overall level is the job of the amplifier, and if it is found impossible to provide adequate level, it is probable that the amplifier is too small for the application. Choose your amplifier carefully, and do not try to compensate for lack of power by using the mixer to increase output level.

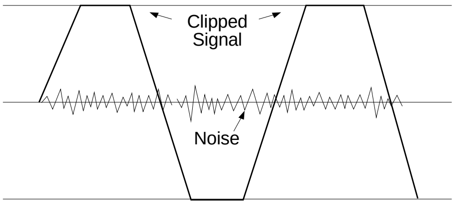
NOTE:

The level of any source signal in the final output is affected by many factors, principally the Input Sensitivity control, Channel Fader, Sub and Mix Faders. You should try to use only as much microphone gain as required to achieve a good balance between signals, with the faders set as described above.

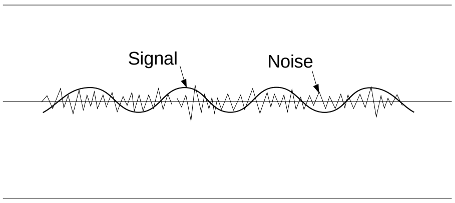
If the input gain is set too high, the channel fader will need to be pulled down too far in compensation to leave enough travel for successful mixing and there is a greater risk of feedback because small fader movements will have a very significant effect on output level. Also there will be a chance of distortion as the signal overloads the channel and causes clipping.

If the gain is set too low, you will not find enough gain on the faders to bring the signal up to an adequate level, and backgound hjiss will be more noticeable.

This is illustrated on the right:



If the signal level is too high, clipping distortion may occur.

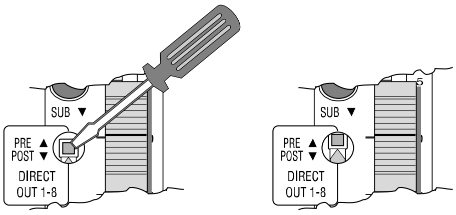


If the signal level is too low it may be masked by the noise.

ADVANCED FEATURES

1 DIRECT OUTPUT

The Direct outputs on the first eight channels allow you to record signals direct to a multitrack tape machine. Normally they are set POST-FADER but can be changed to PRE-FADER by moving the internal PRE/POST switch (13) accessible through an aperture covered by a grommet beside the fader on channel 1 (see below).



POST-FADE PRE-FADE

In the POST-FADER position the outputs may be used as individual effects sends or to provide fader control of recording levels in a studio recording application. For live recording, the outputs would normally be set to PRE-FADE, so that the direct output levels remain unaffected by fader settings for the main PA mix.

The direct output level may be monitored by pressing the PFL switch on the appropriate channel to feed the pre-fade signal to the monitors and the bargraph meters.

In applications where more than 8 tracks are being created, the Direct outputs may be split to feed multiple tape tracks in blocks of 8, and individual tracks enabled on the recorder.

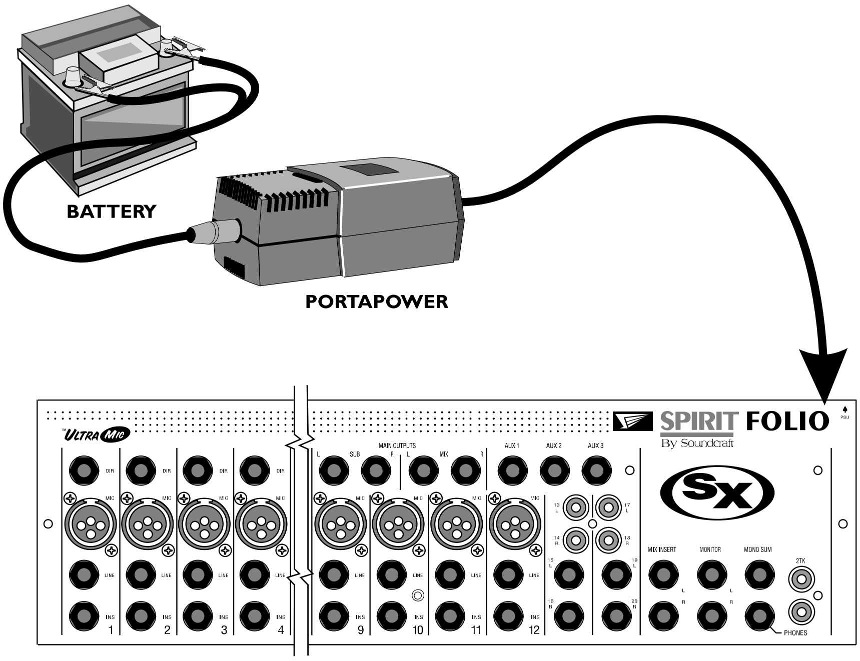
Note also that the insert sends may also be used as an alternative pre-fade, pre-EQ direct output, using a lead with tip and ring shorted together so that the signal path is not interrupted.

Refer to page 9 for an example.

USING THE FOLIO SX ON LOCATION

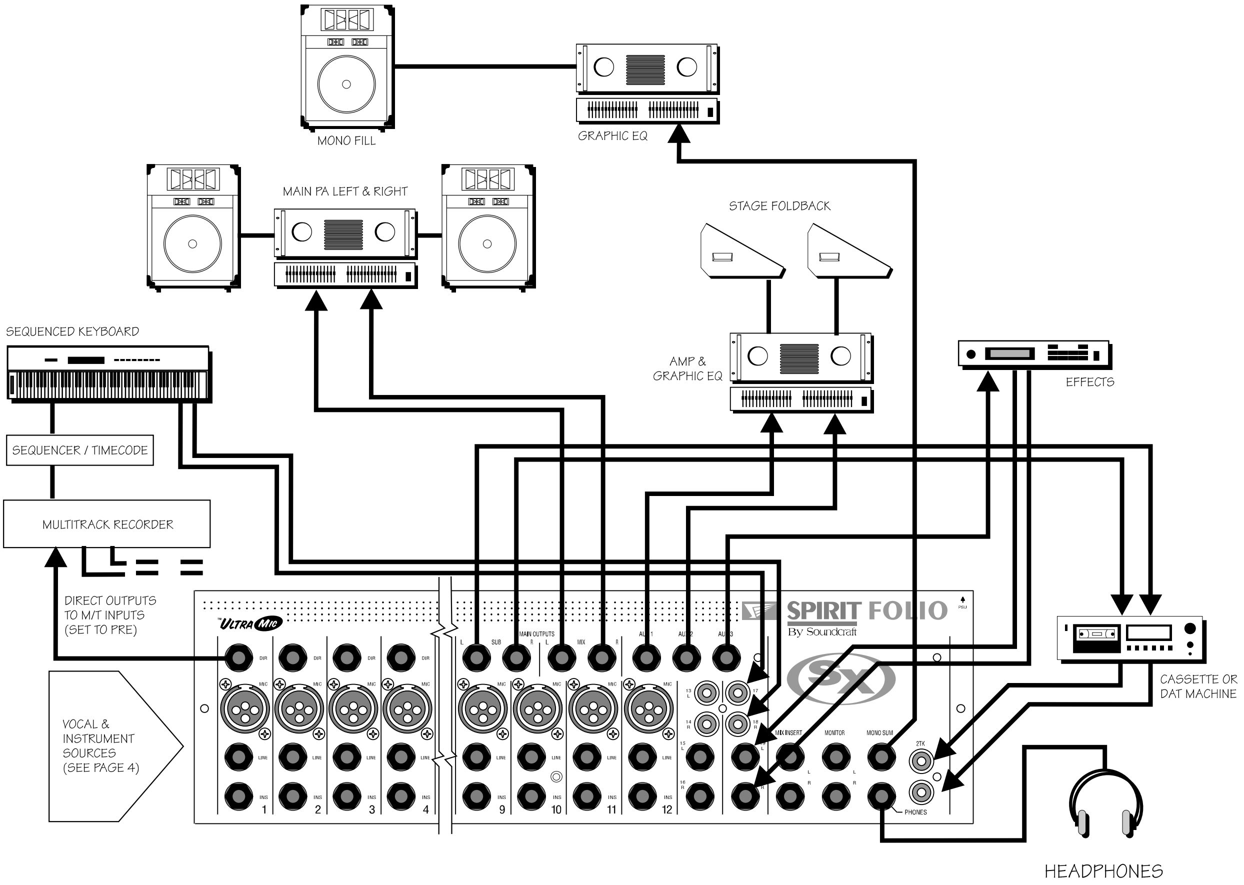
The FOLIO SX may be powered from a DC source of 11-18V using the optional Portapower unit, for applications where a source of mains power is unavailable.

Contact your dealer for further details, quoting Spirit Part No ZZ2849.


BATTERY
PORTAPOWER
SPIRIT FOLIO
By Soundcraft
MAIN OUTPUTS
AUX 1
AUX 2
AUX 3
MIX INSERT
MONITOR
MONO SUM
2TK
PHONES

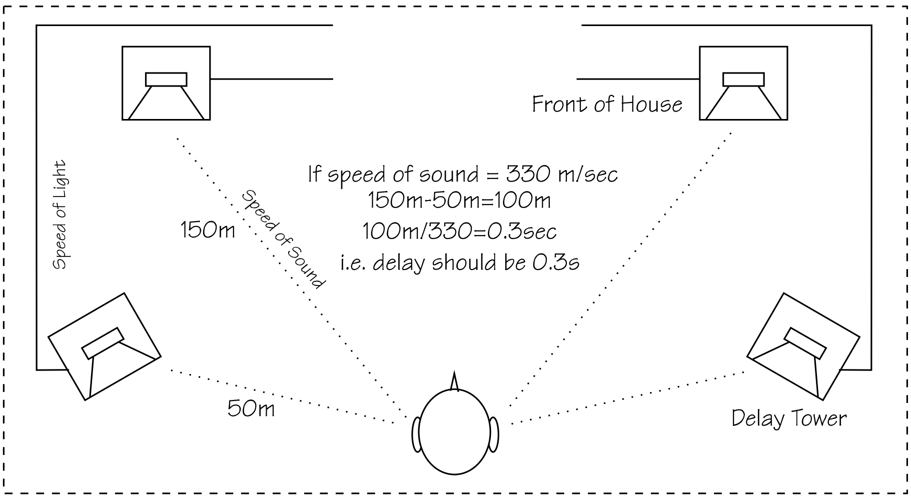
APPLICATION 1 - LIVE SOUND REINFORCEMENT

This drawing shows a typical configuration for sound reinforcement with the main PA fed from Mix L & R, plus an additional centre fill fed from the Mono output. The Aux sends are used for foldback and effects, and Sub L & R are used to feed a recorder. In this case all channels would be routed to SUB, and SUB TO MIX pressed to routed all sources to SUB and MIX.


MONO FILL
GRAPHIC EQ
MAIN PA LEFT & RIGHT
STAGE FOLDBACK
SEQUENCED KEYBOARD
AMP &
GRAPHIC EQ
EFFECTS
SEQUENCER / TIMECODE
MULTITRACK RECORDER
DIRECT OUTPUTS
TO M/T INPUTS
(SET TO PRE)
VOCAL &
INSTRUMENT
SOURCES
(SEE PAGE 4)
UltraMic
SUB
MAIN OUTPUTS
MIX
AUX 1
AUX 2
AUX 3
SPIRIT FOLIO
By Soundcraft
MIX INSERT
MONITOR
MONO SUM
2TK
PHONES
1
2
3
4
9
10
11
12
CASSETTE OR
DAT MACHINE
HEADPHONES

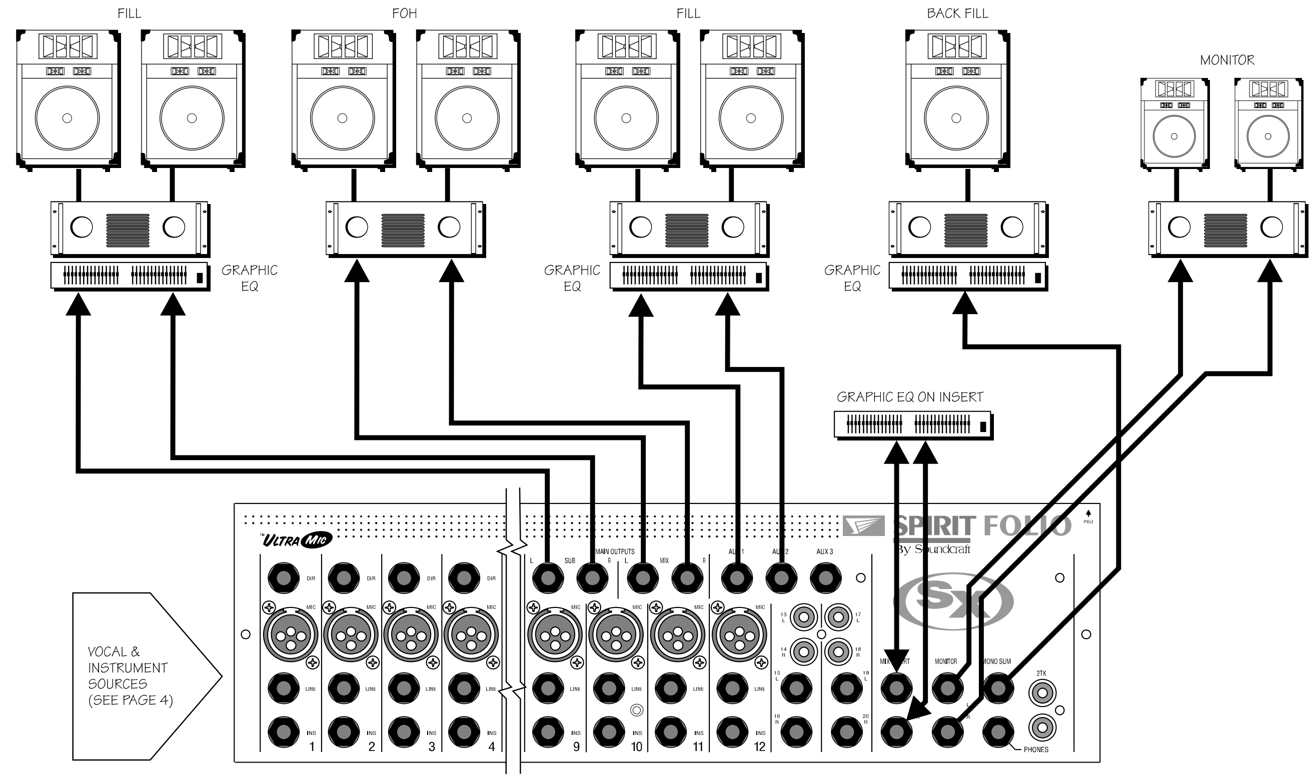
USING DELAY IN REINFORCEMENT SYSTEMS

The drawing below illustrates how to calculate delay settings for fill speakers in multiple speaker installations.


Front of House
If speed of sound = 330 m/sec
150m-50m=100m
100m/330=0.3sec
i.e. delay should be 0.3s
Speed of Light
150m
Speed of Sound
50m
Delay Tower

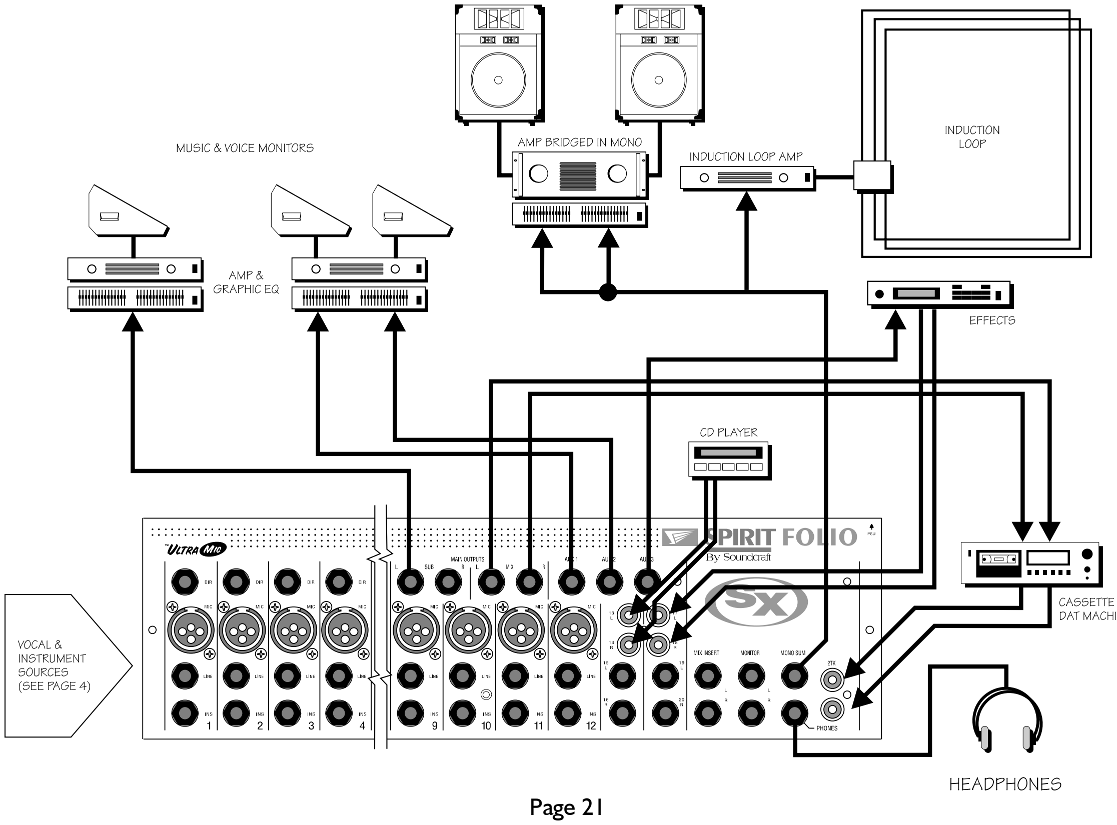
APPLICATION 2 - MULTISPEAKER APPLICATIONS

This configuration demonstrates how multiple speaker configurations can be driven by the FOLIO SX.


FILL
FOH
FILL
BACK FILL
MONITOR
GRAPHIC EQ
GRAPHIC EQ
GRAPHIC EQ
GRAPHIC EQ ON INSERT
SPIRIT FOLIO By Soundcraft
ULTRA MIC
MAIN OUTPUTS
SUB
MIX
AUX 1
AUX 2
AUX 3
MIX INSERT
MONITOR
MONO SUM
2TK
PHONES
DIR
MIC
LINE
INS
1
2
3
4
9
10
11
12
VOCAL & INSTRUMENT SOURCES (SEE PAGE 4)

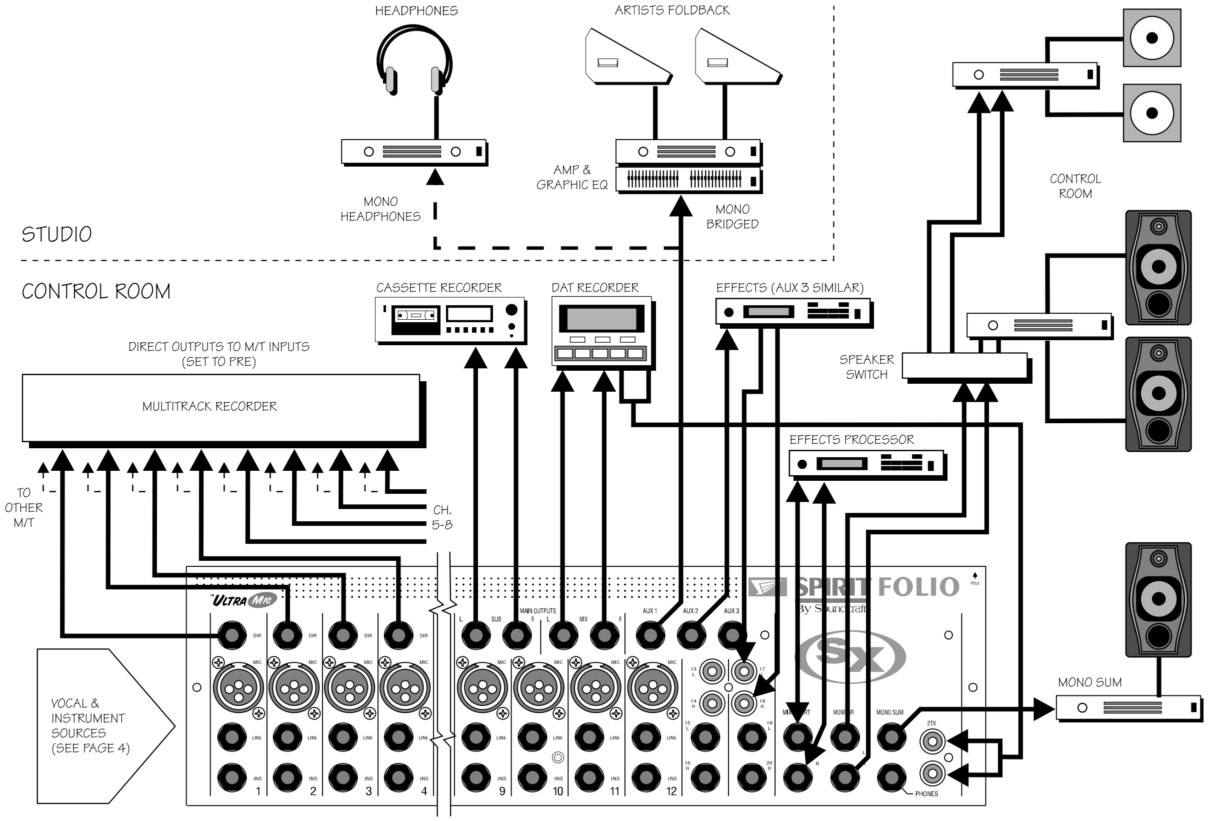
APPLICATION 3 - PLACES OF WORSHIP

This mono configuration uses the Mono output to drive the main speaker system and an induction loop for the hard of hearing. Aux sends are used for monitors and effects and Mix L & R feed a cassette or DAT machine to record the occasion if required.


MUSIC & VOICE MONITORS
AMP & GRAPHIC EQ
AMP BRIDGED IN MONO
INDUCTION LOOP AMP
INDUCTION LOOP
EFFECTS
CD PLAYER
CASSETTE OR DAT MACHINE
HEADPHONES
SPIRIT FOLIO By Soundcraft
ULTRA MIC
MAIN OUTPUTS
SUB
MIX
AUX 1
AUX 2
AUX 3
MIX INSERT
MONITOR
MONO SUM
2TK
PHONES
1
2
3
4
9
10
11
12
VOCAL & INSTRUMENT SOURCES (SEE PAGE 4)

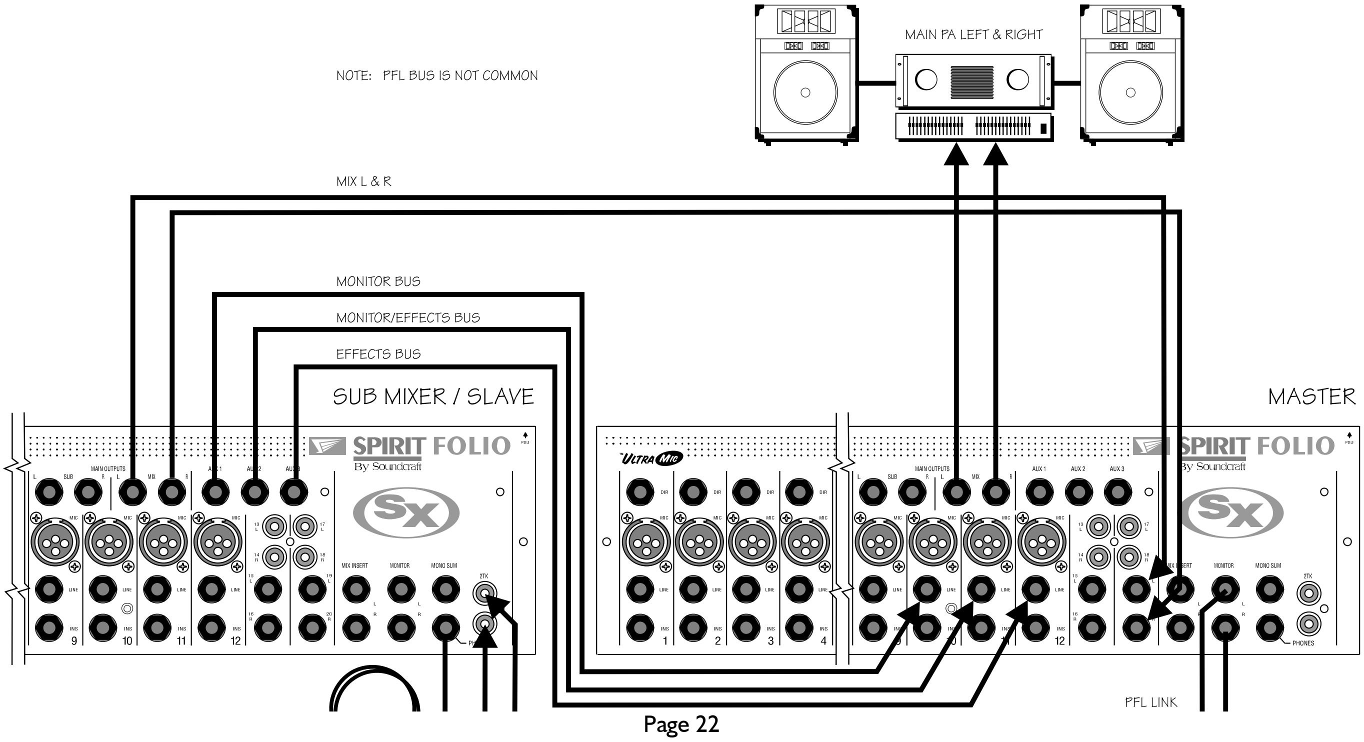
APPLICATION 4 - RECORDING

The direct outputs on channels 1-8 may be used to feed a multitrack recorder as shown. The direct outputs should be set to PRE, so that they are unaffected by fader position. The Mix outputs are used for a preliminary stereo mix on a DAT recorder.


HEADPHONES
ARTISTS FOLDBACK
AMP & GRAPHIC EQ
MONO HEADPHONES
MONO BRIDGED
STUDIO
CONTROL ROOM
CONTROL ROOM
CASSETTE RECORDER
DAT RECORDER
EFFECTS (AUX 3 SIMILAR)
DIRECT OUTPUTS TO M/T INPUTS (SET TO PRE)
MULTITRACK RECORDER
SPEAKER SWITCH
EFFECTS PROCESSOR
TO OTHER M/T
CH. 5-8
VOCAL & INSTRUMENT SOURCES (SEE PAGE 4)
MONO SUM
SPIRIT FOLIO
By Soundcraft
ULTRA MIC
1
2
3
4
9
10
11
12
PHONES

APPLICATION 5 - LINKING TWO FOLIO SX CONSOLES

Two or more FOLIO SX consoles may be combined to create a larger number of input channels by connecting Mix and Aux outputs from one console to the Stereo and Line inputs on another as shown. Note that the PFL/AFL systems remain separate on each console.


MAIN PA LEFT & RIGHT
NOTE: PFL BUS IS NOT COMMON
MIX L & R
MONITOR BUS
MONITOR/EFFECTS BUS
EFFECTS BUS
SUB MIXER / SLAVE
MASTER
SPIRIT FOLIO
By Soundcraft
ULTRA MIC
9
10
11
12
1
2
3
4
PHONES
PFL LINK

CARE OF YOUR MIXER

GENERAL PRECAUTIONS

- Avoid storing or using the mixer in conditions of excessive heat or cold, or in positions where it is likely to be subject to vibration, dust or moisture.
- Keep the mixer clean using a soft dry brush, and an occasional wipe with a damp cloth or ethyl alcohol. Do not use any other solvents which may cause damage to paint or plastic parts.
- Avoid placing drinks or smoking materials on or near the mixer. Sticky drinks and cigarette ash are frequent causes of damage to faders and switches.

  Regular care and inspection will be rewarded by a long life and maximum reliability.

GLOSSARY

AFL (After Fade Listen)	a function that allows the operator to monitor the post-fade signal in a channel independently of the main mix.
auxiliary send	an output from the console comprising a mix of signals from channels and groups derived independently of the main stereo/group mixes. Typically the feeds to the mix are implemented on rotary level controls.
balance	the relative levels of the left and right channels of a stereo signal.
balanced	a method of audio connection which ‘balances’ the signal between two wires and a screen which carries no signal. Any interference is picked up equally by the two wires, but out of phase resulting in cancellation of the interference signal.
clipping	the onset of severe distortion in the signal path, usually caused by the peak signal voltage being limited by the circuit’s power supply voltage.
DAT	Digital Audio Tape, a cassette-based digital recording format.
dB (decibel)	a ratio of two voltages or signal levels, expressed by the equation $dB=20Log_{10} (V_1/V_2)$. Adding the suffix ‘u’ denotes the ratio is relative to 0.775V RMS.
DI(direct injection)/DI Box	the practice of connecting an electric musical instrument directly to the input of the mixing console, rather than to an amplifier and loudspeaker which is covered by a microphone feeding the console.
direct output	a post fade line level output from the input channel, bypassing the summing amplifiers, typically for sending to individual tape tracks during recording.
equaliser	a device that allows the boosting or cutting of selected bands of frequencies in the signal path.
fader	a linear control providing level adjustment
feedback	the ‘howling’ sound caused by bringing a microphone too close to a loudspeaker driven from its amplified signal.
foldback	a feed sent back to the artistes via loudspeakers or headphones to enable them to monitor the sounds they are producing.
frequency response	the variation in gain of a device with frequency.
gain/input sensitivity	the variation in level of the signal

(sub) group	an output into which a group of signals can be mixed.
headroom	the available signal range above the nominal level before clipping occurs.
highpass filter	a filter that rejects low frequencies.
impedance balancing	a technique used on unbalanced outputs to minimise the effect of hum and interference when connecting to external balanced inputs.
insert	a break point in the signal path to allow the connection of external devices, for instance signal processors or to another mixer line level signals at a nominal level of -10 to +6dBu, usually coming from a low impedance source.
pan (pot)	abbreviation of 'panorama': controls levels sent to left and right outputs.
peaking	an equaliser response curve affecting only a band of frequencies i.e. based on a bandpass response.
PFL (pre-fade listen)	a function that allows the operator to monitor the pre-fade signal in a channel independently of the main mix.
phase	a term used to describe the relationship of two audio signals. In-phase signals reinforce each other, out-of-phase signals result in cancellation.
polarity	a term used to describe the orientation of the positive and negative poles of an audio connection. Normally connections are made with positive to positive, negative to negative and this would correct polarity. If this is reversed, the result will be out-of-phase signals (see 'phase' above).
post-fade	the point in the signal path after the monitor or master fader and therefore affected by fader position.
pre-fade	the point in the signal path before the monitor or master fader position and therefore unaffected by the fader position.
rolloff	a fall in gain at the extremes of the frequency response.
shelving	an equaliser response affecting all frequencies above or below the break frequency i.e. a highpass or lowpass derived response.
spill	acoustic interference from other sources.
transient	a momentary rise in the signal level.
unbalanced	a method of audio connection which uses a single wire and the cable screen as the signal return. This method does not provide the noise immunity of a *balanced* input (see above)
+48V	the phantom power supply, available at the channel mic inputs, for condenser microphones and active DI boxes.

TYPICAL SPECIFICATIONS

NOISE

Measured RMS, 20Hz to 20kHz Bandwidth
Line inputs selected at unity gain and terminated 150R

AUX, MIX, SUB O/PS	16 Inputs routed, faders & pots down	<85 dBu
E.I.N.	Microphone Input, Maximum Gain, terminated 150R	-129 dBu

CROSSTALK

	1kHz	*10kHz*
Channel mute	<95 dB	<85 dB
Fader cutoff (rel to 0 mark)	<90dB	<80dB
Routing isolation	<90dB	<80dB
Aux Sends pots offness	<85dB	<80dB

FREQUENCY RESPONSE

Mic/Line In to any output 25Hz to 30kHz	<1dB

T.H.D.	Mic Sens. -30dBu, +20dBu at all outputs @ 1kHz	< 0.006%

INPUT & OUTPUT IMPEDANCES

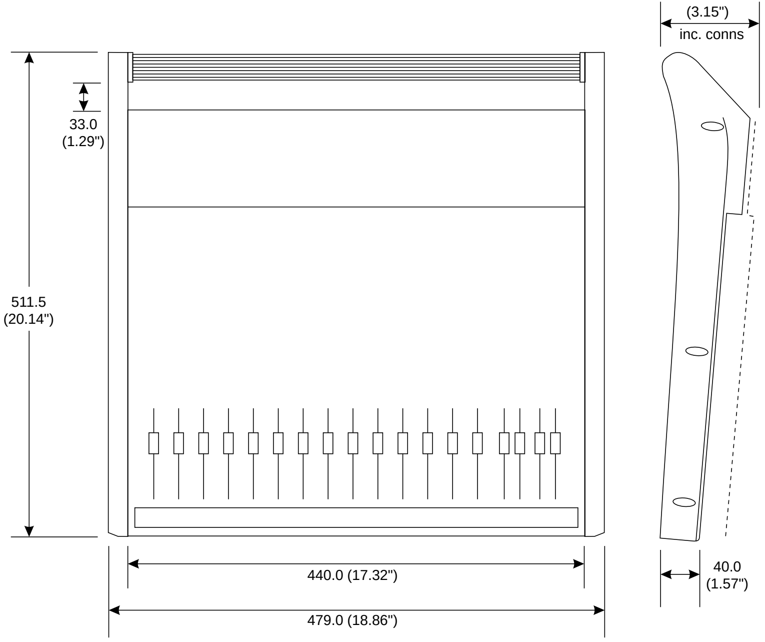
Microphone Input	1.8 kΩ
Line Input	10 kΩ
Stereo Input A/B	12 kΩ
Stereo Input C/D	10 kΩ
Direct Out, Sub-Mix, Aux, Insert	75 Ω

INPUT & OUTPUT LEVELS

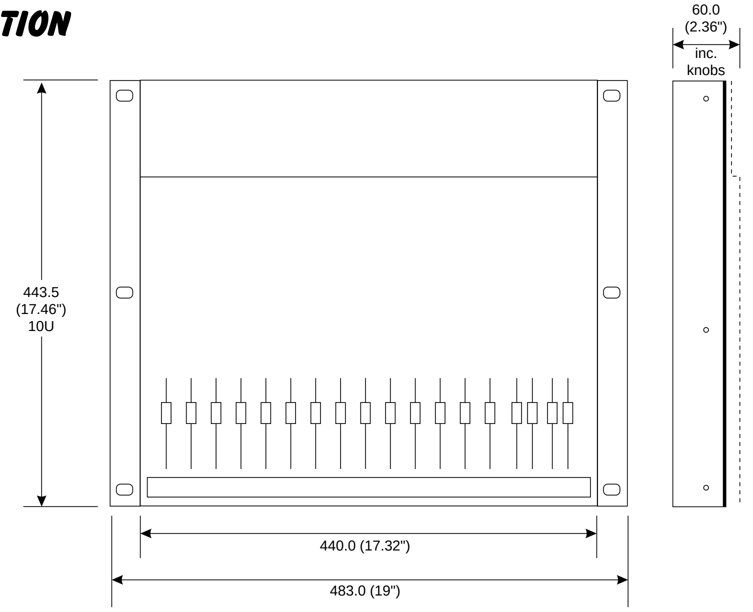
Mic Input Maximum Level	+22 dBu
Line Input Maximum Level	>+30dBu
Stereo Input Maximum Level	>+30 dBu
Headphones (@200Ω)	150mW

DIMENSIONS

All dimensions are in millimetres (inches in brackets).


80.0
(3.15")
inc. conns
33.0
(1.29")
511.5
(20.14")
40.0
(1.57")
440.0 (17.32")
479.0 (18.86")

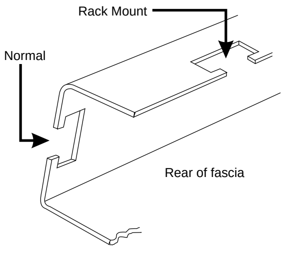
RACK MOUNT OPTION


60.0
(2.36")
inc.
knobs
443.5
(17.46")
10U
440.0 (17.32")
483.0 (19")

FITTING THE RACK MOUNT OPTION

The FOLIO SX may be rack mounted using the optional rack-mounting kit which requires only a posidriv screwdriver and pliers for installation. The procedure is as follows:

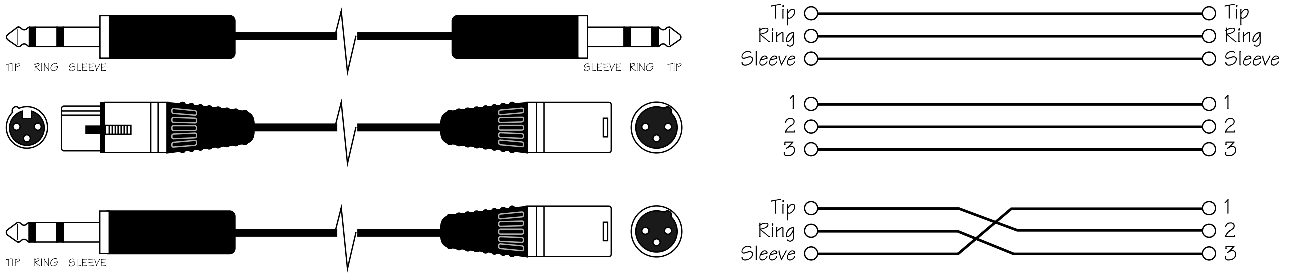
- ☐ Remove the moulded side cheeks by releasing 3 screws each side.
- ☐ Remove the base panel by releasing 10 screws, noting the small cutout at the front edge.
- ☐ Carefully release the power connector by sqeezing the moulded side clips with pliers (note that you do not have to disconnect the wiring) and reposition it in the cutout on the bottom edge of the panel (see illustration).
- ☐ Replace the base panel with the cutout at the rear of the desk, positioned over the power connector.
- ☐ Fit the rack-mount side brackets with the mounting flanges to the front of the desk.


Rack Mount
Normal
Rear of fascia

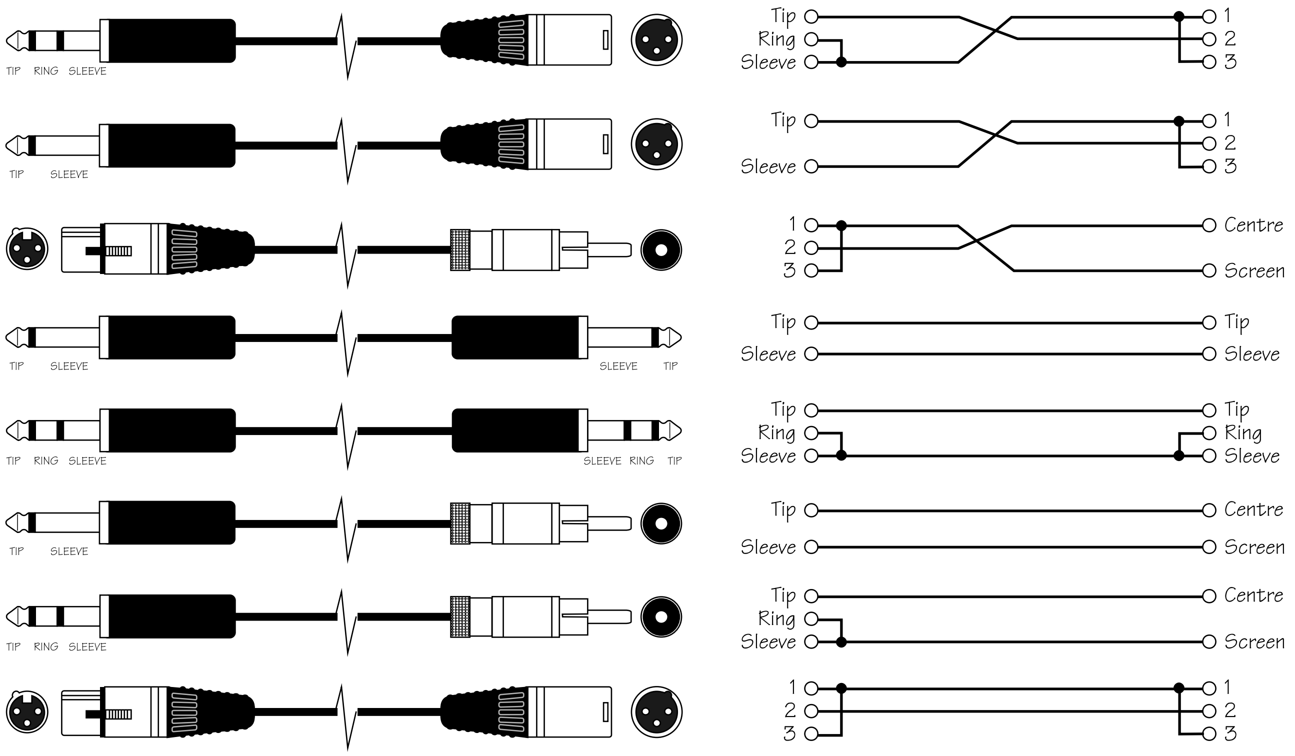
Power Supply Connector Location

APPENDIX 1 - TYPICAL CONNECTING LEADS

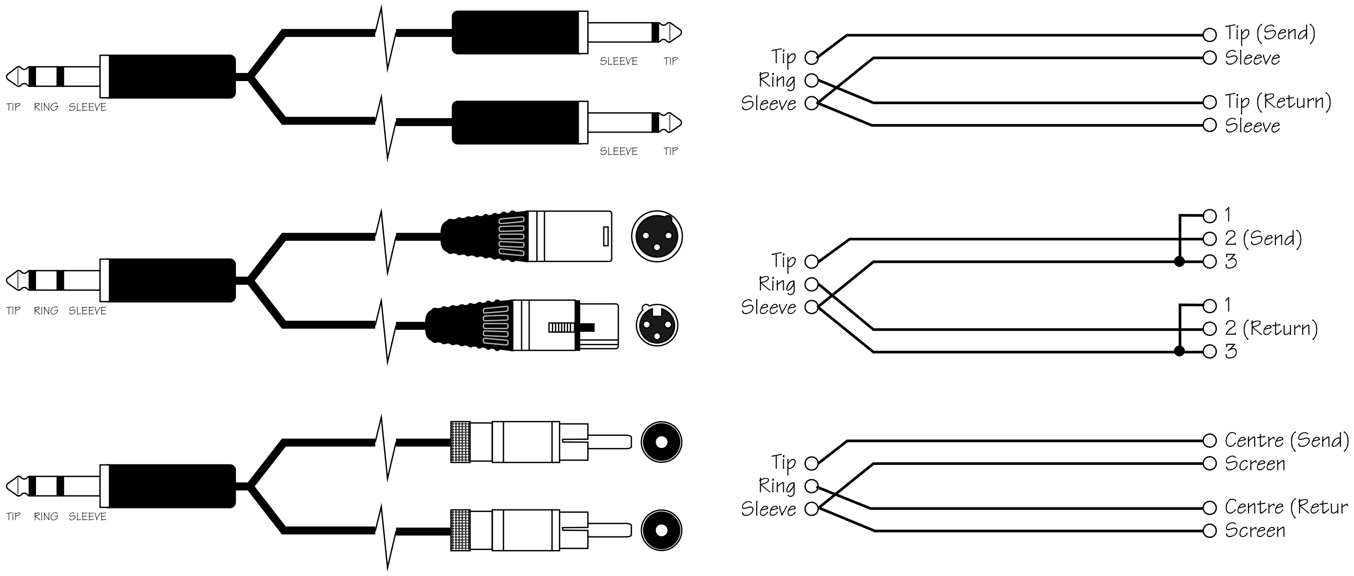
Balanced - Line, Mix L & R, Sub L & R, Aux, Stereo Inputs


TIP RING SLEEVE
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
1
2
3
1
2
3
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2
3

Unbalanced - Direct Output, Control Room L & R, Stereo Inputs

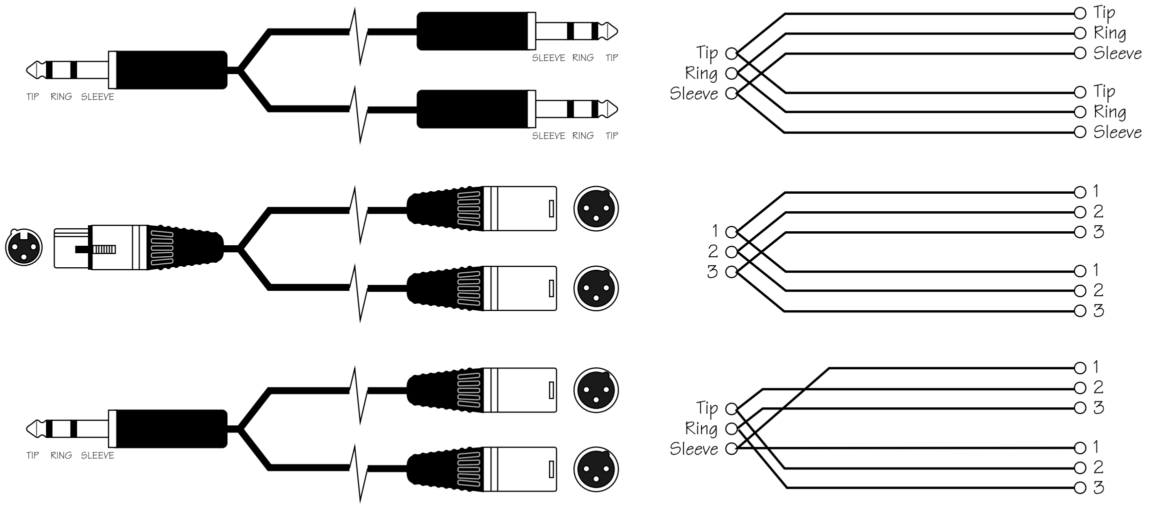

TIP RING SLEEVE
Tip
Ring
Sleeve
1
2
3
TIP SLEEVE
Tip
Sleeve
1
2
3
1
2
3
Centre
Screen
TIP SLEEVE
SLEEVE TIP
Tip
Sleeve
Tip
Sleeve
TIP RING SLEEVE
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
TIP SLEEVE
Tip
Sleeve
Centre
Screen
TIP RING SLEEVE
Tip
Ring
Sleeve
Centre
Screen
1
2
3
1
2
3

Insert Leads - Mono Inserts, Mix L & R Inserts

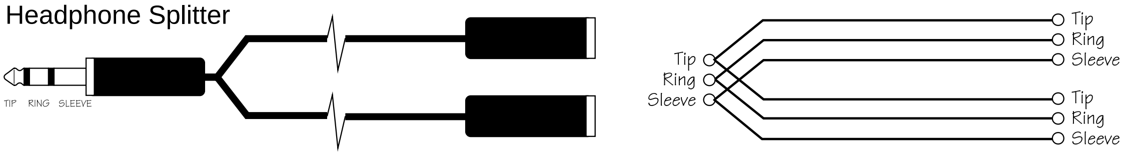

SLEEVE TIP
TIP RING SLEEVE
SLEEVE TIP
Tip
Ring
Sleeve
Tip (Send)
Sleeve
Tip (Return)
Sleeve
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2 (Send)
3
1
2 (Return)
3
TIP RING SLEEVE
Tip
Ring
Sleeve
Centre (Send)
Screen
Centre (Retur
Screen

'Y' Leads (Balanced)

Where used....Aux, Mix outputs

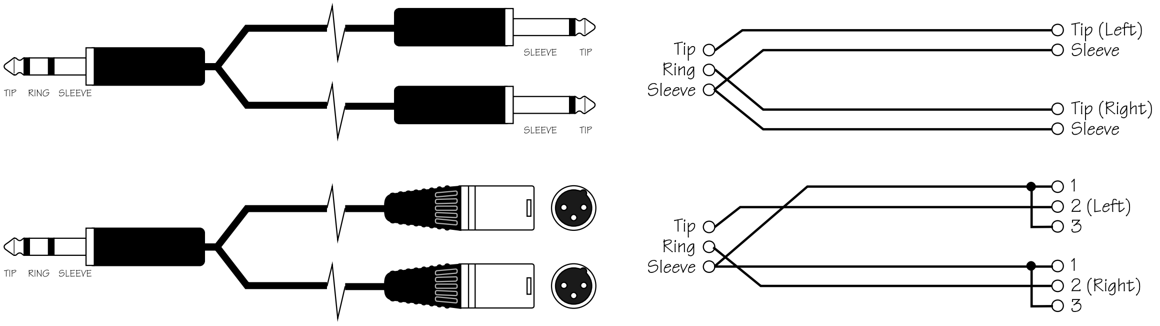

TIP RING SLEEVE
SLEEVE RING TIP
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
Tip
Ring
Sleeve
1
2
3
1
2
3
1
2
3
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2
3
1
2
3

Headphone Splitter

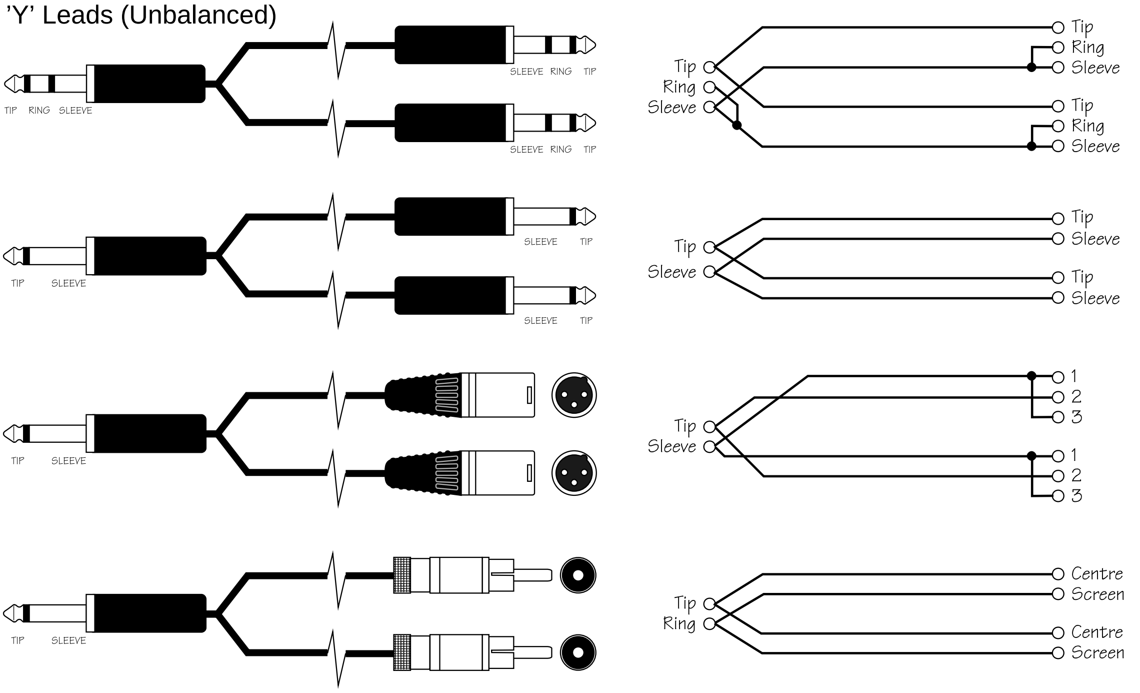

TIP RING SLEEVE
Tip
Ring
Sleeve
Tip
Ring
Sleeve
Tip
Ring
Sleeve

Headphone Separator

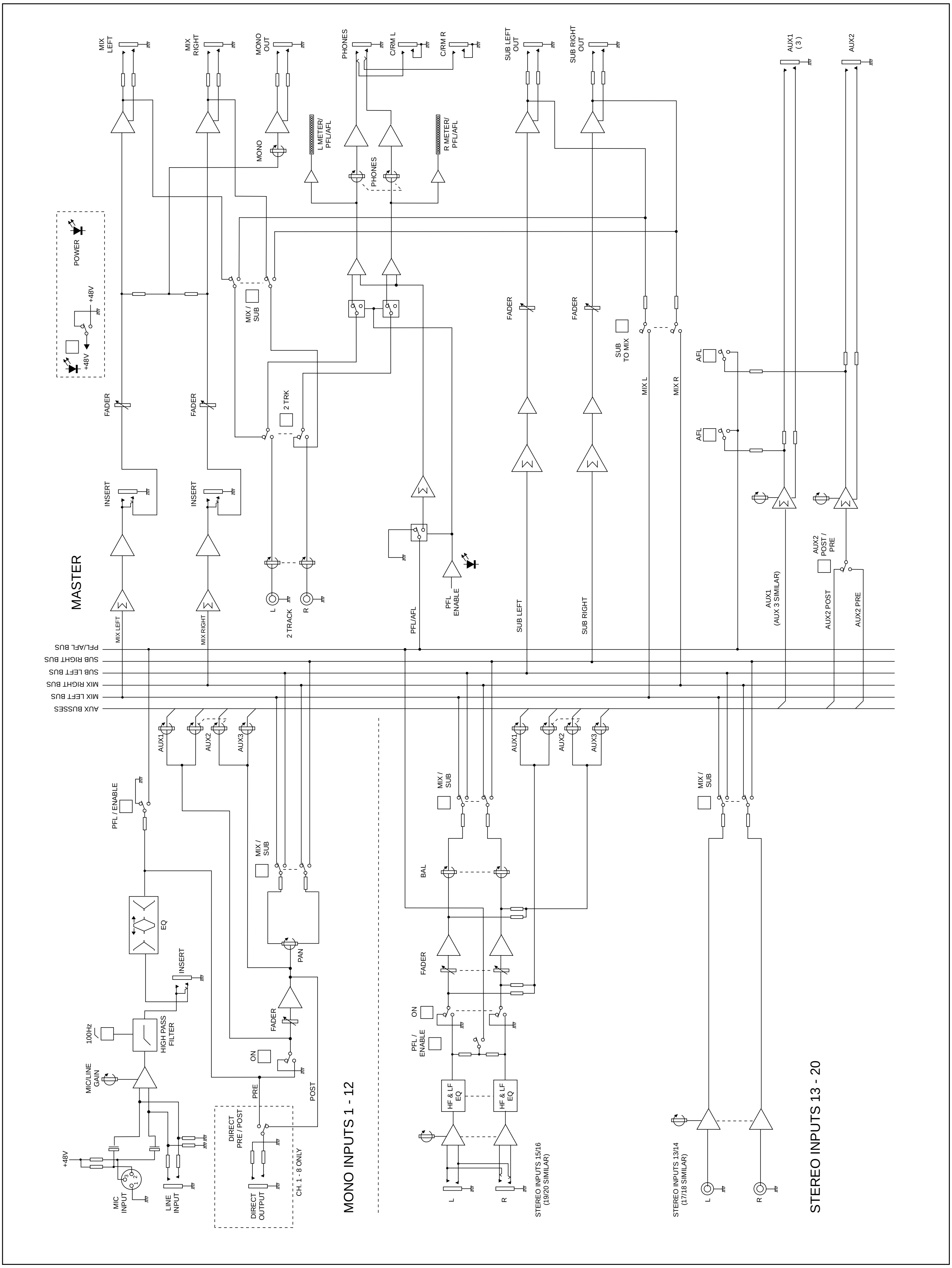
Note: for every doubling of headphones the load impedance is halved. Do not go below 200R.

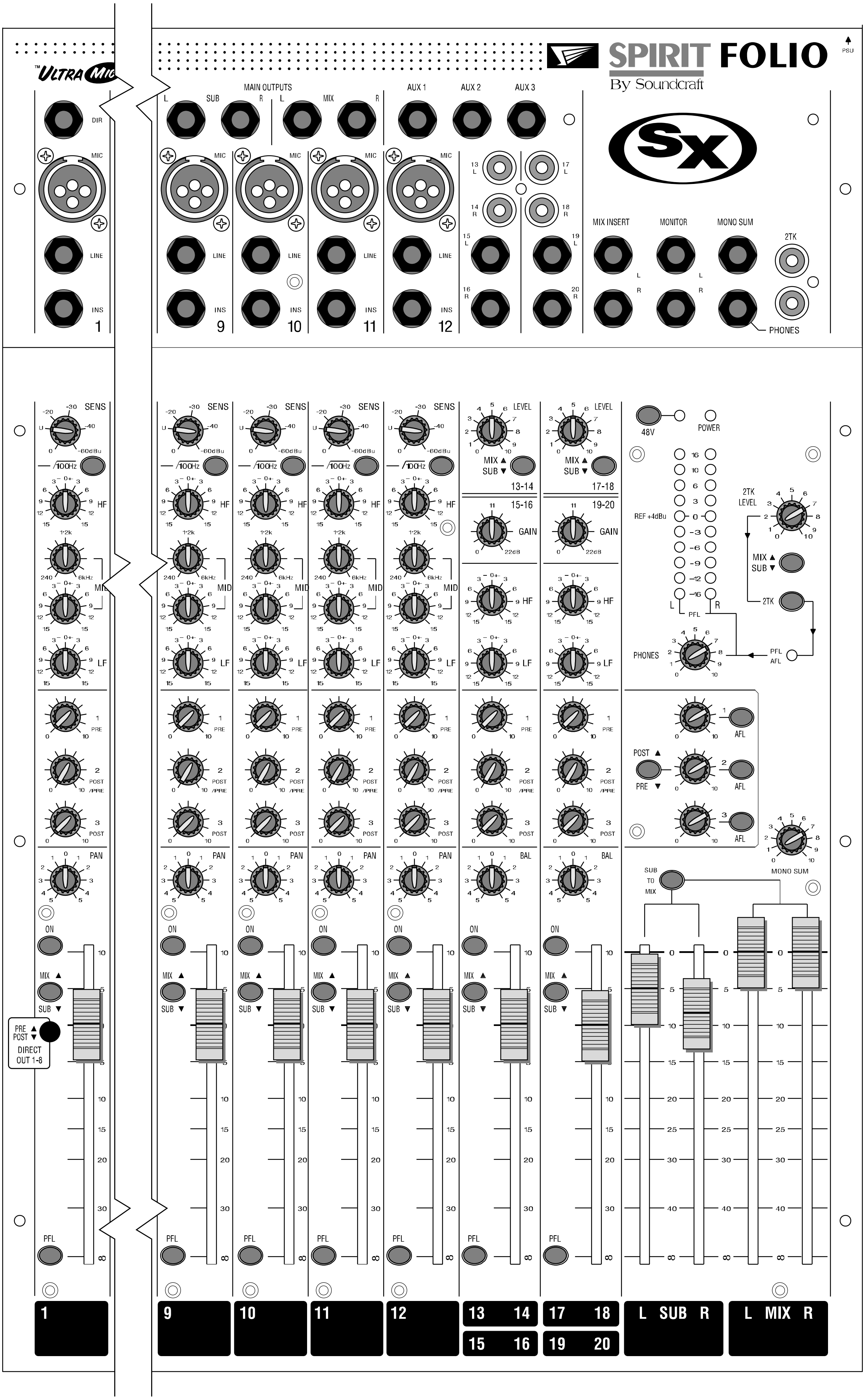

TIP RING SLEEVE
SLEEVE TIP
SLEEVE TIP
Tip
Ring
Sleeve
Tip (Left)
Sleeve
Tip (Right)
Sleeve
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2 (Left)
3
1
2 (Right)
3

'Y' Leads (Unbalanced)


TIP RING SLEEVE
SLEEVE RING TIP
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
Tip
Ring
Sleeve
TIP SLEEVE
SLEEVE TIP
SLEEVE TIP
Tip
Sleeve
Tip
Sleeve
Tip
Sleeve
TIP SLEEVE
Tip
Sleeve
1
2
3
1
2
3
TIP SLEEVE
Tip
Ring
Centre
Screen
Centre
Screen

SYSTEM BLOCK DIAGRAM


MIX LEFT
MIX RIGHT
MONO OUT
PHONES
C/RM L
C/RM R
SUB LEFT OUT
SUB RIGHT OUT
AUX1 (3)
AUX2
MONO
L METER/ PFL/AFL
R METER/ PFL/AFL
PHONES
POWER
+48V
+48V
MIX / SUB
FADER
FADER
SUB TO MIX
MIX L
MIX R
AFL
AFL
FADER
FADER
2 TRK
INSERT
INSERT
MASTER
L
R
2 TRACK
PFL ENABLE
AUX2 POST / PRE
AUX1 (AUX 3 SIMILAR)
AUX2 POST
AUX2 PRE
MIX LEFT
MIX RIGHT
PFL/AFL
SUB LEFT
SUB RIGHT
PFL/AFL BUS
SUB RIGHT BUS
SUB LEFT BUS
MIX RIGHT BUS
MIX LEFT BUS
AUX BUSSES
AUX1
AUX2
AUX3
AUX1
AUX2
AUX3
PFL / ENABLE
MIX / SUB
MIX / SUB
MIX / SUB
BAL
EQ
INSERT
PAN
FADER
FADER
ON
ON
PFL / ENABLE
100Hz
HIGH PASS FILTER
MIC/LINE GAIN
HF & LF EQ
HF & LF EQ
PRE
POST
DIRECT PRE / POST
+48V
MIC INPUT
LINE INPUT
DIRECT OUTPUT
CH. 1 - 8 ONLY
MONO INPUTS 1 - 12
L
R
STEREO INPUTS 15/16 (19/20 SIMILAR)
STEREO INPUTS 13/14 (17/18 SIMILAR)
L
R
STEREO INPUTS 13 - 20


SPIRIT FOLIO
By Soundcraft
PSU
ULTRA MIC
MAIN OUTPUTS
L SUB R
L MIX R
AUX 1
AUX 2
AUX 3
DIR
MIC
LINE
INS
1
9
10
11
12
13 L
14 R
17 L
18 R
15 L
16 R
19 L
20 R
MIX INSERT
MONITOR
MONO SUM
2TK
PHONES
SENS
100Hz
HF
MID
LF
PRE
POST
/PRE
PAN
BAL
LEVEL
GAIN
MIX ▲
SUB ▼
13-14
15-16
17-18
19-20
48V
POWER
REF +4dBu
PFL
2TK LEVEL
AFL
POST ▲
PRE ▼
MONO SUM
SUB TO MIX
ON
PFL
PRE ▲
POST ▼
DIRECT OUT 1-8
1
9
10
11
12
13 14
15 16
17 18
19 20
L SUB R
L MIX R

CE

This equipment complies
with the EMC Directive
89/336/EEC

For further details contact the address below

SPIRIT
By Soundcraft

Spirit By Soundcraft™
Harman International Industries Ltd., Cranborne House, Cranborne Industrial Estate,
Cranborne Road, Potters Bar, Herts., EN6 3JN U.K.
Tel: +44 (0) 1707 665000   Fax: +44 (0) 1707 665461
http://www.spirit-by-soundcraft.co.uk

H A Harman International Company

Part No: ZM0161

Soundcraft

SPIRIT LX7

Contents

Introduction **2**
  Einleitung *2*
  Introduction *3*
  Introduzione *3*
  Introducción *3*
Getting Started **4**
  Übersicht *4*
  Démarrage *4*
  Per Iniziare *4*
  Puesta en Marcha *4*
Using the Folio LX7 . . . . **6**
  Vorbereitungen Bedienelemente des LX7 . . . . *6*
  Utilisation de la LX7 . . . . *7*
  Utilizzo del Mixer LX7 . . . . *7*
  Usando la LX7 . . . . *7*
Setting Up & Troubleshooting **20**
  Richtiger Einsatz und Fehlersuche *20*
  Réglages et problèmes de fonctionnement *21*
  Set Up & Guida all'Individuazione dei Problemi . . *21*
  Ajustes y Solución de Problemas *21*
Applications **22**
  Anwendungen *22*
  Applications *22*
  Applicazioni *22*
  Aplicaciones *22*
System Block Diagram **27**
  Block Diagramm *27*
  Synoptique *27*
  Diagramma a blocchi *27*
  Diagrama de Bloques del Sistema *27*
Technical Information **28**
  Technische Informationen *28*
  Informations Techniques *28*
  Informazioni Tecniche *28*
  Información Técnica *28*

USERS GUIDE

Handbuch
Manuel d'utilisation
Manuele Utente
Guía del usuario

Featuring ™ ULTRA MIC

For your own safety and to avoid invalidation of the warranty please read this section carefully.

Zu Ihrer eigenen Sicherheit und damit die Garantie-Ansprüche gewahrt bleiben, lesen Sie bitte diesen Abschnitt sehr sorgfältig durch!

Pour votre sécurité et pour éviter d'annuler la garantie, lisez attentivement cette section.

Per la vostra sicurezza e per non invalidare la garanzia leggette con la massima attenzione i paragrafi seguenti.

Para su propia seguridad y para evitar la invalidación de la garantía, por favor, lea cuidadosamente esta sección.

SAFETY PRECAUTIONS

WARNING: THIS UNIT MUST BE EARTHED
Under no circumstances should the mains earth be disconnected from the mains lead.

The wires in the mains lead are coloured in accordance with the following code:

Earth: Green and Yellow (Green/Yellow - US)

Neutral: Blue (White - US)

Live: Brown (Black - US)

As the colours of the wires in the mains lead may not correspond with the coloured markings identifying the terminals in your plug, proceed as follows:

- The wire which is coloured Green and Yellow must be connected to the terminal in the plug which is marked with the letter E or by the earth symbol.
- The wire which is coloured Blue must be connected to the terminal in the plug which is marked with the letter N.
- The wire which is coloured Brown must be connected to the terminal in the plug which is marked with the letter L.

Ensure that these colour codings are followed carefully in the event of the plug being changed.

This unit is capable of operating over a range of mains voltages by means of a 4-position mains input fuse carrier. It is important to ensure that the correct voltage setting is selected for the level of local mains voltage supply, for safe, uninterrupted operation. Use a small screwdriver to prise the fuse carrier from its location in the connector.

To avoid the risk of fire, replace the mains fuse only with the correct value fuse, as marked on the rear panel.

The internal power supply unit contains no user serviceable parts. Refer all servicing to a qualified service engineer, through the appropriate Spirit dealer.

SICHERHEITSHINWEISE

WARNUNG: DAS MISCHLPULT MUSS GEERDET SEIN.

Unter keinen Umständen darf die Erdleitung von den Hauptanschlüssen getrennt werden!

Die Verdrahtung der Hauptleitungen stimmen mit folgenden Farbcodes überein:

● GRÜN / GELB oder Markierung E:	Gehäuseschutzleiter
● BLAU oder Markierung N:	Mittelpunktsleiter
● BRAUN oder Markierung L:	Phase

Achten Sie auch bei einem Steckdosenwechsel auf die Farbcodierung an den Steckkontakten der Steckdose.

Das Mischpult kann mit 4 unterschiedlichen Netzspannungen betrieben werden. Es ist daher sehr wichtig, daß die eingestellte Netzspannung mit der an der Steckdose anliegenden Netzspannung übereinstimmt. Eine falsch eingestellte Netzspannung kann zu einem Defekt am Mischpult führen. Mit einem kleinen Schraubenzieher wird der Sicherungsträger in richtige Netzposition gebracht.

Um die Gefahr eines Elektrobrandes zu verhindern, muß die zuersetzende Hauptsicherung die auf dem Gerätegehäuse angegebenen Absicherungswerte haben.

Das eingebaute Netzgerät enthält keine Komponenten, die vom Benutzer gewaretet werden müssen. Eventuelle Serviceleistungen dürfen nur von qualifiziertem Fachpersonal oder dem SPIRIT-Fachmann durchgeführt werden.

PRECAUTIONS

CETTE APPAREIL DOIT IMPÉRATIVEMENT ÊTRE RELIÉ À LA TERRE.

En aucune circonstance, la terre ne doit etre déconnectée du câble secteur.

Les conducteurs du câble secteur sont identifiés comme suit :

Vert/Jaune	Terre
Bleu	Neutre
Brun	Phase

Pour éviter tout risque d'incendie, remplacez le fusible uniquement par un fusible de la valeur correcte indiquée sur l'alimentation.

L'alimentation interne ne contient pas de pièces accessibles par l'utilisateur.

Référez-vous à du personnel qualifié.

NORME DI SICUREZZA

ATTENZIONE: QUESTO APPARECCHIO VA COLLEGATO A TERRA

In nessun caso il collegamento di terra va staccato da quello dell'alimentazione.

I cavi sono colorati come segue:

Terra: Verde e Giallo
Neutro: Blu
Vivo: Marrone

In caso i colori siano diversi operare come segue:

- Cavo Verde e Giallo va collegato al terminale E o con il simbolo di terra.
- Cavo Blu va collegato al terminale N.
- Cavo Marrone va collegato al terminale L.

Assicurarsi che questi codici-colore siano rispettati in caso di sostituzione della presa.

Questo apparecchio può operare con diverse tensioni grazie al portafusibile regolabile su 4 posizioni. E' importante assicurarsi che la tensione selezionata sia la stessa della rete a cui ci si collega. Per spostare il portafusibile dal suo alloggiamento usare un cacciavite.

Per evitare il rischio di incendio sostituire il fusibile solo con uno dello stesso valore, indicato sul pannello posteriore.

L'alimentatore interno contiene parti non riparabili dall'utente. Rivolgersi ad un centro assistenza autorizzato.

PRECAUCIONES DE SEGURIDAD

ATENCION: ESTA UNIDAD DEBE ESTAR CONECTADA A TIERRA

Bajo ninguna circunstancia la toma de tierra debe ser desconectada del cable de alimentación principal

Los conductores en el terminal de red se encuentran codificados por colores del siguiente modo:

Tierra: Verde y Amarillo
Neutro: Azul
Vivo: Marrón

En caso de que los colores de los conductores del terminal de red no coincidan con los colores de las marcas que identifican los terminales en su enchufe, proceda del siguiente modo:

- El conductor de color verde y amarillo debe conectarse al terminal del enchufe que este marcado con la letra E o por el símbolo de tierra.
- El conductor de color azul debe conectarse al terminal del enchufe que este marcado con la letra N, o sea, de color negro.
- El conductor de color marrón debe conectarse al terminal del enchufe que este marcado con la letra L, o sea, de color rojo.

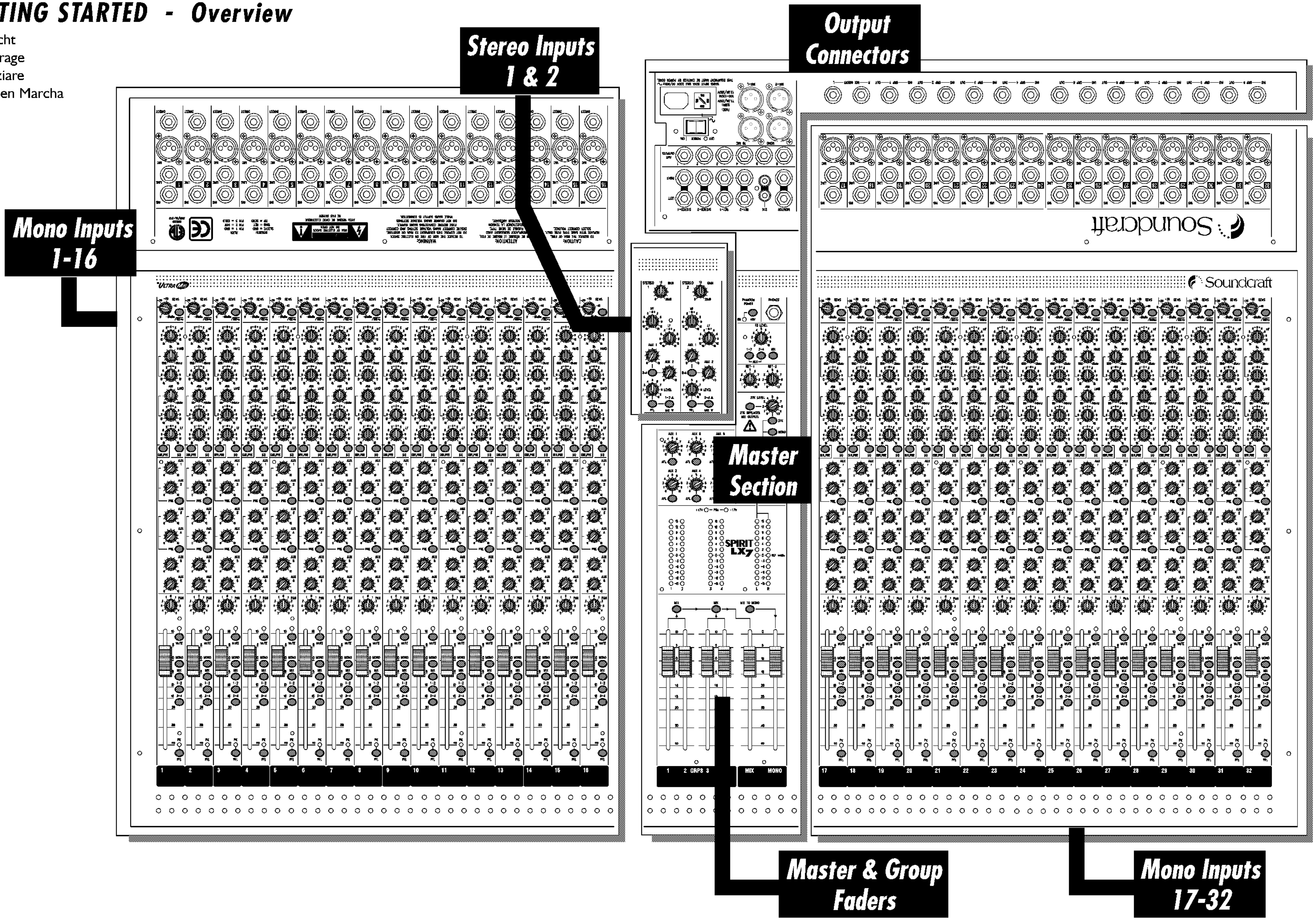
Asegúrese de seguir cuidadosamente este código de colores en caso de que deba sustituirse el enchufe.
Esta unidad es capaz de operar sobre una amplia gama de tensiones gracias a un soporte del fusible de red de 4 posiciones. Es importante asegurar que la selección de tensión es adecuada al nivel del suministro de red local para una operación segura e ininterrumpida. Use un pequeño destornillador para ajustar la posición del soporte del fusible en el conector.

Para evitar riesgos de incendio, sustituya el fusible sólo con otro del mismo valor, como se indica en el panel trasero.

La fuente de alimentación interna contiene partes no reemplazables por el usuario. Para reparaciones diríjase a un servicio técnico oficial, a través de su distribuidor Spirit.

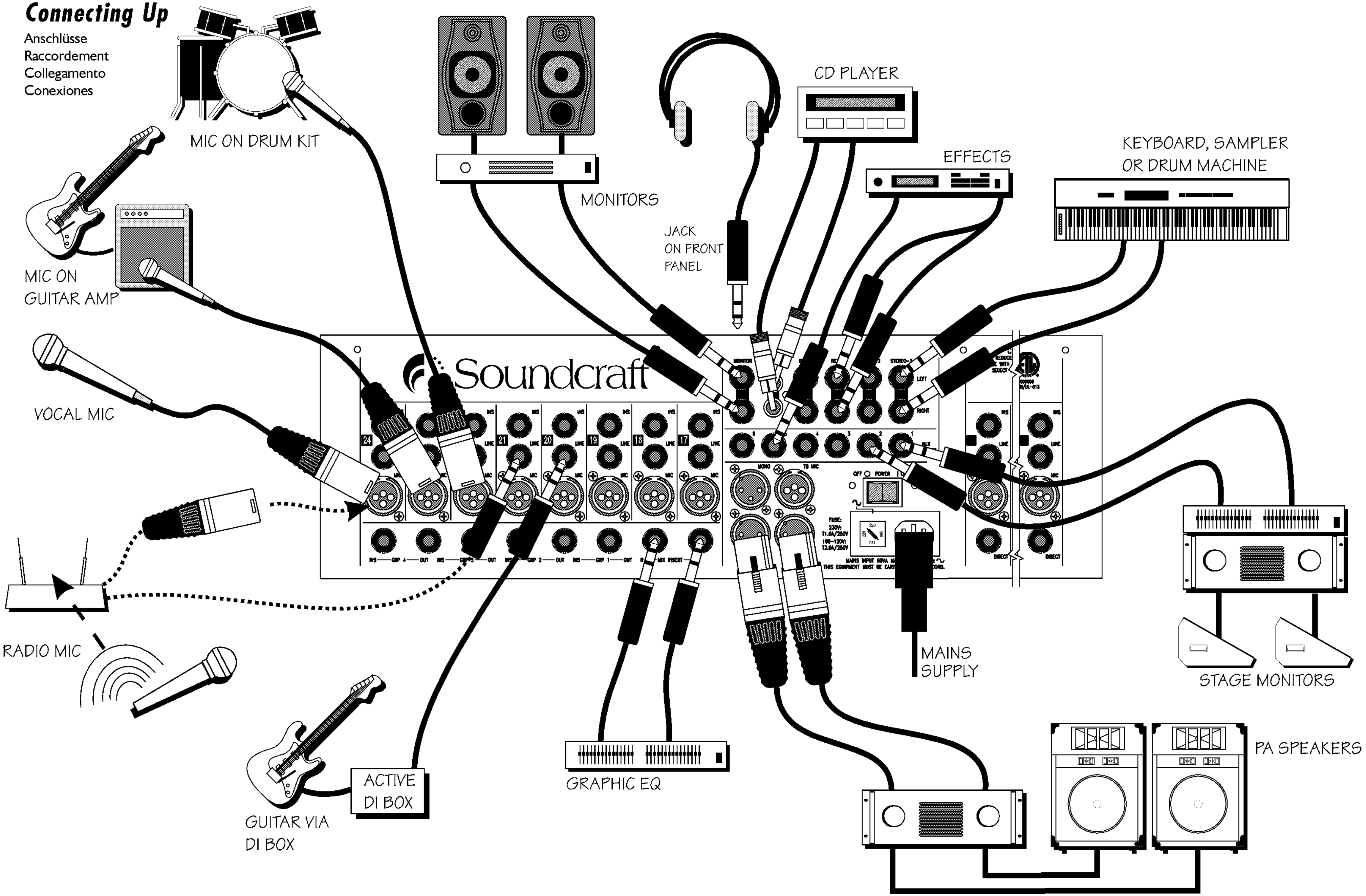
GETTING STARTED - Overview

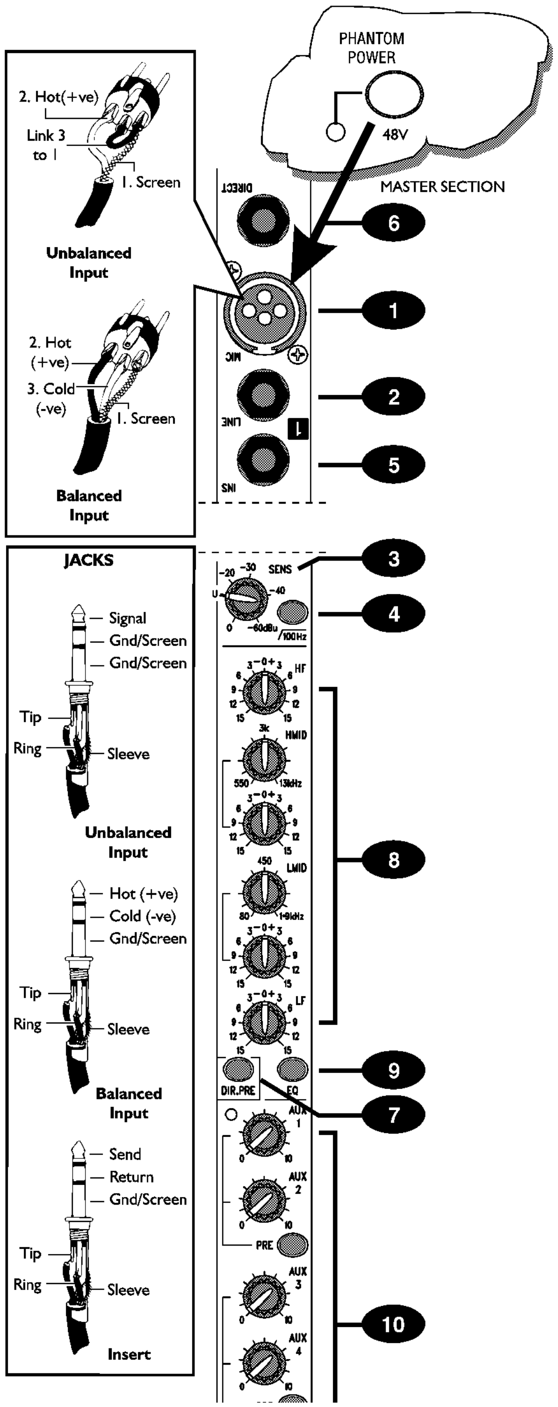
Übersicht
Démarrage
Per Iniziare
Puesta en Marcha


Output Connectors
Stereo Inputs 1 & 2
Mono Inputs 1-16
Master Section
Master & Group Faders
Mono Inputs 17-32
Soundcraft
SPIRIT LX7

Connecting Up

Anschlüsse
Raccordement
Collegamento
Conexiones


MIC ON DRUM KIT
MONITORS
JACK
ON FRONT
PANEL
CD PLAYER
EFFECTS
KEYBOARD, SAMPLER
OR DRUM MACHINE
MIC ON
GUITAR AMP
VOCAL MIC
Soundcraft
RADIO MIC
STAGE MONITORS
MAINS
SUPPLY
GRAPHIC EQ
PA SPEAKERS
ACTIVE
DI BOX
GUITAR VIA
DI BOX



USING LX7

Mono Input Channel

1 MIC INPUT ™ULTRA MIC

The mic input accepts XLR-type connectors and is designed to suit a wide range of BALANCED or UNBALANCED signals. Professional dynamic, condenser or ribbon mics are best because these will be LOW IMPEDANCE. You can use low-cost HIGH IMPEDANCE mics, but the level of background noise will be higher. If you press the 48V switch down (beside the phones jack) the socket provides a suitable powering voltage for professional condenser mics (this is also known as Phantom Power).

ONLY connect condenser microphones with the 48V powering OFF (switch UP), and ONLY turn the 48V powering on or off with all output faders DOWN, to prevent damage to the mixer or external devices.

TAKE CARE when using unbalanced sources, which may be damaged by the phantom power voltage on pins 2 & 3 of the XLR connector.

Unplug any mics if you want to use the LINE Input. The input level is set using the SENS knob.

2 LINE INPUT

Accepts 3-pole `A' gauge (TRS) jacks. Use this high impedance input for sources other than mics, such as keyboards, drum machines, synths, tape machines or guitars. The input is BALANCED for low noise and top quality from professional equipment, but you can use UNBALANCED sources by wiring up the jacks as shown, although you should then keep cable lengths as short as possible. Unplug anything in the MIC input if you want to use this socket. Set the input level using the SENS knob.

3 SENS (Sensitivity)

This knob sets how much of the source signal is sent to the rest of the mixer. Too high, and the signal will distort as it overloads the channel. Too low, and the level of any background hiss will be more noticeable and you may not be able to get enough signal level to the output of the mixer.
Setting the knob to the `U' mark gives unity gain for the LINE input. Note that some sound equipment, particularly that intended for domestic use, operates at a lower level (-10dBV) than professional equipment and will therefore need a higher gain setting to give the same output level.
See `Setting Up & Troubleshooting' on page 20 to learn how to set SENS correctly.

4 100Hz HI-PASS FILTER

Pressing this switch activates a steep 18dB per octave filter which reduces the level of bass frequencies only. Use this in live PA situations to clean up the mix, reducing stage rumble or 'popping' from microphones.

VORBEREITUNGEN BEDIENELEMENTE DES LX7

Mono-Eingangs-Kanal

1 MIC-EINGANG ™ULTRA MIC

Der XLR-Mic-Eingang bietet die Möglichkeit, symmetrische und unsymmetrische Geräte anzuschließen. Professionelle, dynamische oder Grenzflächen-Mikrofone sind am geeignetsten, da diese eine niedrige Impedanz haben. Es können auch preisgünstige Mikrofone mit hoher Impedanz verwendet werden. Hier ist jedoch das Hintergrund-Rauschen höher. Wird der 48V-Schalter neben der Kopfhörer-Buchse gedrückt, wird die sogenannte Phantom-Spannung für Kondensator-Mikrofone eingeschaltet.

Es dürfen nur Kondensator-Mikrofone mit der Phantom-Spannung betrieben werden. Weiterhin dürfen nur bei ausgeschaltetem 48V-Schalter die Kondensator-Mikrofone angeschlossen werden. Den 48V-Schalter nur betätigen, wenn alle Ausgangsfader heruntergepegelt sind. Hierdurch werden Defekte am Mixer und an externen Geräten vermieden.

Vorsicht bei unsymmetrischen Quellen, da diese durch die 48V-Spannung an den Pins 2 und 3 des XLR-Anschlusses zerstört werden könnten.

Bevor Sie den Line-Eingang benutzen, müssen Sie das Mikrofon aus den XLR-Buchse ausstecken.

2 LINE-EINGANG

Am LINE-Eingang werden 6,3 mm Klinkenstecker angeschlossen. Dieser Eingang mit hoher Impedanz ist für Quellen, wie Keyboards, Drum-Machines, Synthesizer, Band-Maschinen oder Gitarren vorgesehen. Der Eingang ist symmetrisch ausgelegt, mit niedrigem Rauschpegel und besitzt bei Einsatz professionellen Equipments hervorragende technische Eigenschaften. Es können jedoch auch unsymmetrische Quellen angeschlossen werden. Hierbei müssen die Klinkenstecker wie gezeigt verlötet sein. Achten Sie jedoch darauf, daß die Kabellänge so kurz wie möglich ist. Bevor Sie den Line-Eingang benutzen, müssen Sie das Mikrofon aus den XLR-Buchse ausstecken. Mit dem SENS-Regler wird die Eingangsempfindlichkeit eingestellt.

3 SENS (Empfindlichkeit)

Mit diesem Regler wird der Signalpegel des Eingangs eingestellt. Ist das Signal zu hoch eingestellt, werden Signal-Verzerrungen entstehen. Ist das Signal zu niedrig eingestellt, entsteht ein hörbares Hintergrundrauschen und der Signal-Pegel am Ausgang des Mixers ist zu schwach. Wird der Regler auf 'U' gestellt, ist eine Vorverstärkung von =1 für den LINE-Eingang eingestellt. Es gibt Sound-Equipment, daß für den häuslichen Bereich konstruiert ist: Dieses Equipment hat einen niedrigen Pegel (-10 dBV) als professionelles Equipment und benötigt daher eine höhere Einstellung der Vorverstärkung, um den gleichen Ausgangspegel zu erreichen. Im Kapitel 'Richtiger Einsatz und Fehlersuche' sind wichtige Hinweise für die Einstellung des SENS-Reglers gegeben.

4 100 Hz-HOCHPASSFILTER

Durch Drücken des 100 Hz-Schalters wird ein Filter mit einer Flankensteilheit von 18 dB/Oktave geschaltet. Dieses Filter reduziert nur den Pegel der Bass-Frequenzen. In Live PA-Situationen werden z.B. Bühnenschwingungen oder 'Mikrofon-Poppen' reduziert.

UTILISATION DE LA LX7

Voie d'Entrée Mono

❶ ENTRÉE MICRO ULTRA MIC ™UltraMic

L'entrée micro sur connecteur de type XLR est conçue pour convenir à une large gamme de signaux SYMETRIQUES ou ASYMETRIQUES. Les micros professionnels dynamiques, statiques ou à ruban sont les meilleurs parce qu'ils sont à BASSE IMPEDANCE. Vous pouvez utiliser des micros peu coûteux à HAUTE IMPEDANCE, mais le niveau du bruit de fond sera plus haut. Si vous mettez l'alimentation FANTOME en service à l'aide de la touche stiuée à côté de la prise casque, l'embase fournit la tension appropriée pour des micros professionnels statiques.

Utilisez l'alimentation fantôme 48V SEULEMENT avec des microphones statiques. Mettez l'alimentation fantôme 48V en ou hors service avec tous les faders EN BAS, pour éviter des dommages à la console ou aux equipements externes.

FAITES ATTENTION lors d'utilisation de sources asymétriques, elles peuvent être endommagées par la tension de l'alimentation fantôme présente sur les broches 2 et 3 du connecteur XLR.

Débranchez le micro si vous voulez utiliser l'entrée LIGNE. Le gain d'entrée est réglé par le potentiomètre SENS.

❷ ENTREE LIGNE

Cette entrée sur jack 6.35 STEREO est destinée à des sources autres que des micros, telles que des claviers, des boîtes à rythmes, des synthétiseurs, des magnétophones ou des guitares. L'entrée est SYMETRIQUE pour un bon rapport signal/bruit et pour du matériel professionnel, mais vous pouvez utiliser des sources ASYMETRIQUES en câblant les jacks comme montré ; dans ce cas utilisez des câbles aussi courts que possible. Débranchez tout micro de l'entrée MICRO si vous voulez utiliser cette embase. Le gain d'entrée est réglé par le potentiomètre SENS.

❸ SENS (Sensibilité)

Ce potentiomètre permet d'adapter la sensibilité du préampli micro au niveau de signal de la source. Réglé trop haut, le signal sera distordu et surchargera la voie. Trop bas, le niveau de bruit sera plus audible et vous ne pourrez pas obtenir assez de niveau en sortie de console.Le gain unitaire de l'entrée LIGNE est à la position U. Notez que certains matériels audio, en
particulier ceux destinés à une utilisation domestique, fonctionnent à un niveau plus bas (-10dBV) que le matériel professionnel et auront donc besoin d'un gain plus élevé pour donner le même niveau de sortie.

Voir Réglages et problèmes de fonctionnement à la page 9 pour apprendre comment régler le potentiomètre SENS correctement.

❹ FILTRE PASSE-HAUT A 100Hz

Cette touche insère un filtre à 18dB par octave qui réduit le niveau des basses fréquences indésirables. A utiliser en sonorisation pour réduire les bruits de scènes ou le «pop» des microphones.

UTILIZZO DEL MIXER

Ingresso Microfonico Mono

❶ INGRESSO MIC ™UltraMic

Accetta connettori di tipo XLR ed è progettato per una vasta gamma di segnali a basso livello BILANCIATI e SBILANCIATI. I microfoni professionali dinamici, a condensatore o a nastro sono i migliori perchè saranno a BASSA IMPEDENZA. Mentre usando un microfono economico ad ALTA IMPEDENZA non si ottiene lo stesso grado di immunità alle interferenze sul cavo microfonico e quindi il livello del rumore di fondo potrebbe essere alto. Se viene inserita l'ALIMENTAZIONE PHANTOM, la presa fornisce la tensione adatta ai microfoni professionali a condensatore.

Collegare i microfoni a condensatore solo con l'alimentazione Phantom disinserita ed attivarla solo con tutti i fader abbassati per evitare danni alla consolle o ad apparecchi esterni.

Fare attenzione con le sorgenti sbilanciate, che potrebbero essere danneggiate dall'alimentazione Phantom sui pin 2&3 del contatto XLR.

Staccare tutti i microfoni in caso di utilizzo dell'ingresso line. Il livello di ingresso viene regolato con la manopola SENS.

❷ INGRESSO LINE

Accetta jack a 3-poli 'A'. Usate questo ingresso per sorgenti che non siano microfoni, ad esempio tastiere, drum machines, synth, registratori o chitarre. L'ingresso è BILANCIATO per un baso rumore e protezione da interferenze, ma è possibile usare sorgenti SBILANCIATE collegando i jack come mostrato in figura, sebbene dobbiate mantenere la lunghezza dei cavi il più corto possibile. Impostate il livello di ingresso con la manopola SENS.

❸ SENS

Questa manopola imposta la quantità di segnale da mandare al resto del mixer. Segnali troppo alti saranno distorti. Segnali troppo bassi faranno risaltare il fruscio di sottofondo ed il livello di uscita dal mixer non sarà sufficiente.

Fermando la manopola sul segno "U" si avrà il gain per l'ingresso LINE. Alcuni apparecchi, specialmente quelli per uso domestico, operano ad un livello più basso (-10dBV) di quelli professionali quindi avranno bisogno di un gain maggiore per raggiungere lo stesso livello.

Vd. "SetUp Iniziale" per impostare correttamente la manopola SENS.

❹ FILTRO PASSA-ALTO 100Hz

Premendo questo tasto si attiva un filtro di 18dB per ottava che riduce solo il livello delle basse frequenze ed è una caratteristica eccezionale. Da usare in situazioni PA live per un mix più pulito, riducendo il rumore del palco o il "popping" dei microfoni.

USANDO LA LX7

Canal de Entrada Mono

❶ ENTRADA MIC ™UltraMic

La entrada Mic acepta conectores tipo XLR y ha sido diseñada para acomodarse a un amplio abanico de señales balanceadas y no balanceadas. Los micrófonos profesionales dinámicos, de condensador o de cinta son los más adecuados ya que presentan una baja impedancia. Si utiliza micrófonos baratos de alta impedancia, el nivel del ruido de fondo puede ser mayor. Si activa la alimentación phantom de 48V (al lado del conector jack de auriculares), el conector suministrará una tensión de alimentación adecuada para los micrófonos profesionales de condensador (la cual es conocida como alimentación Phantom).

SOLO conecte micrófonos de condensador cuando la alimentación phantom esté desactivada (conmutador arriba), y SOLO active la alimentación phantom con todos los faders de salida ABAJO, para evitar daños en el mezclador o en aparatos externos.

TENGA CUIDADO cuando use fuentes no balanceadas, ya que pueden ser dañadas por la tensión que aparece sobre los pins 2 y 3 del conector XLR.

Desconecte cualquier micrófono si desea utilizar la entrada de LINE. El nivel de entrada se ajusta a través del control SENS.

❷ ENTRADA DE LINEA

Acepta jacks de 1/4" (TRS) de 3 polos. Utilice esta entrada de alta impedancia para otras fuentes que no sean micrófonos como, por ejemplo, teclados, cajas de ritmos, sintetizadores, magnetófonos o guitarras. Esta entrada es BALANCEADA para ofrecer una bajo nivel de ruido e inmunidad frente a las interferencias, sin embargo, puede utilizar fuentes no balanceadas cableando los conectores tal como se muestra, aunque deberá mantener la longitud de los cables tan corta como le sea posible. Observe que el anillo debe ser "masa" si la fuente es no balanceada. El nivel de entrada se ajusta a través del control SENS. Desconecte cualquier conexión MIC cuando use la entrada LINEA del mismo canal.

❸ SENS (Sensibilidad)

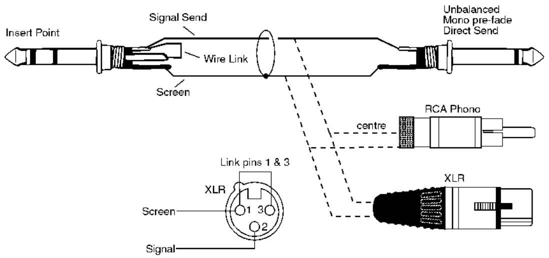
Este botón regula la cantidad de señal fuente enviada al resto del mezclador. Si se sube demasiado, la señal distorsionará al saturar el canal. Si se baja demasiado, el nivel del ruido de fondo será más apreciable y no tendrá suficiente nivel a la salida del mezclador.

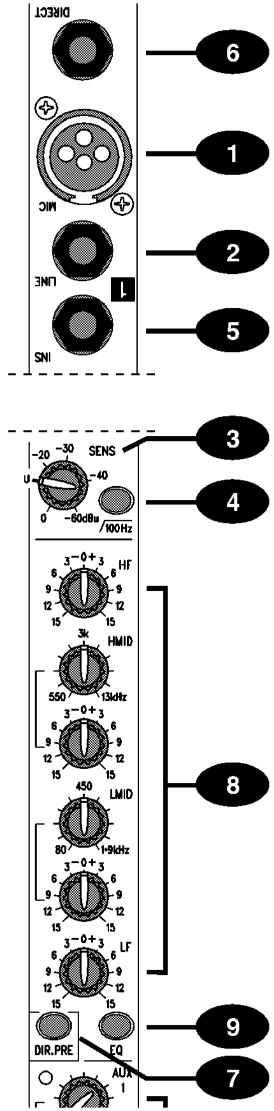
Con el control en la marca U se aplica ganancia unidad para la entrada de línea. Algunos equipos, sobre todo los domésticos, operan a niveles más bajos (-10dBV) que los profesionales, y por lo tanto requieren mayor ganancia para ofrecer el mismo nivel de salida.

Vea "Configuración y Solución de Problemas" en la página 9 para aprender cómo ajustar correctamente SENS.

❹ FILTRO PASA ALTO A 100 HZ

Pulsando este conmutador se activa un filtro de 18dB por octava que reduce el nivel solamente de las bajas frecuencias. Utilícelo en sonorizaciones en directo para reducir las vibraciones procedentes del escenario o los sonidos oclusivos que puedan producirse en los micrófonos





5 INSERT POINT

The unbalanced, **pre-EQ** insert point is a break in the channel signal path, allowing limiters, compressors, special EQ or other signal processing units to be added in the signal path. The Insert is a 3-pole 'A' gauge jack socket which is normally bypassed. When a jack is inserted, the signal path is broken, just before the EQ section. The Send may be tapped off as an alternative pre-fade, pre-EQ direct output if required, using a lead with tip and ring shorted together so that the signal path is not interrupted.

6 DIRECT OUTPUT

Channels 1-16 have a dedicated Direct Output which allows direct connection to external devices, for example to feed Tape Machines or effects units. The pre-fade direct output level may be monitored by pressing the PFL switch on the appropriate channel to feed the pre-fade signal to the monitors and the bargraph meters.

7 DIR. PRE

The Direct Outputs on channels 1-16 are normally set POST-FADER for use as effects sends or to provide fader control of recording levels in a studio recording application. For live recording the outputs can be individually changed to PRE-FADER by pressing this switch, so that the direct output level remains unaffected by fader settings for the main PA mix.

8 EQUALISER

The Equaliser (EQ) allows fine manipulation of the frequency bands, and is particularly useful for improving the sound in live PA applications where the original signal is often far from ideal and where slight boosting or cutting of particular voice frequencies can really make a difference to clarity.

HF EQ

Turn clockwise to boost high (treble) frequencies (12kHz and above) by up to 15dB, adding crispness to cymbals, vocals and electronic instruments. Turn anticlockwise to cut by up to 15dB, reducing hiss or excessive sibilance which can occur with certain types of microphone. Set the knob in the centre-detented position when not required.

MID EQ (HMID & LMID)

There are two pairs of knobs which work together to form HI and LO MID frequency EQ sections. The lower knob in each pair provides 15dB of boost and cut, just like the HF EQ knob, but the frequency at which this occurs can be set by the upper knob over a range of 550Hz to 13kHz (HMID) or 80Hz to 1.9kHz (LMID). This allows some truly creative improvement of the signal in live situations, because the mid bands cover the range of most vocals. Listen carefully as you use these controls together to find how particular characteristics of, for instance, a vocal signal can be enhanced or reduced. Set the gain (lower) knob to the centre-detented position when not required. Note: Q is set at 1.5.

LF EQ

Turn clockwise to boost low (bass) frequencies (60Hz and below) by up to 15dB, adding warmth to vocals or extra punch to synths, guitars and drums. Turn anticlockwise to cut low frequencies by up to 15dB for reducing hum, stage rumble or to improve a mushy sound. Set the knob to the centre-detented position when not required.

5 INSERT (Einschleifpunkt)

Der unsymmetrische, pre EQ Insert ist eine Unterbrechung im Signalweg eines Kanals. An diesem Punkt können Limiter, Kompressoren, spezielle EQs oder andere Signal-Prozessoren in den Signalweg eingeschleift werden. Der Insert-Eingang ist eine 6,3 mm Buchse, die das Kanalsignal normalerweise umgeht, da sie bypass geschaltet ist. Wird ein Klinkenstecker in die Buchse gesteckt, so wird der Signalweg vor der EQ-Sektion unterbrochen. Der Send des Inserts kann als pre-fade, pre-EQ Direkt-Ausgang verwendet werden. Hierzu muß im Klinkenstecker auf jeden Fall auch eine elektrische Verbindung zwischen dem Tip und dem Ring des Steckers gelötet werden, damit der Signalweg nicht unterbrochen wird.

6 DIRECT OUTPUT (Direkt-Ausspielweg)

Die Kanäle 1-16 besitzen Direkt-Ausspielwege, um externe Geräte, wie z.B. Aufnahme- oder Effektgeräte direkt ansteuern zu können. Durch Drücken der PFL-Taste kann der pre-fade Pegelausgang auf dem entsprechenden Kanal über die Monitor-Ausgänge abgehört oder auf den Aussteuerungsanzeige abgelesen werden.

7 DIR.PRE

Die Direct-Ausgänge der Kanäle 1-16 sind normalerweise post-fade geschaltet und damit zum Einsatz als Effect Sends oder zum Einstellen des Aufnahmepegels in einer Studio-Anwendung vorgesehen. Für Live-Aufnahmen können die einzelnen Direct-Ausgänge durch den DIR.PRE-Schalter auf pre-fade umgeschaltet werden. So sind die Direct-Ausgänge von den Fader-Einstellungen des PA-Mixers unbeeinflußt.

8 EQUALIZER

Der Equalizer (EQ) ermöglicht eine exakte Einstellung der Frequenz-Bänder und eignet sich besonders zur Verbesserung des Live-PA-Sounds. Gerade beim Live-Sound wird der Klang des Originalsignals z.B. aufgrund der örtlichen Gegebenheiten oft nicht erreicht, da z.B. nur geringfügiges Verstärken oder Abschwächen der Gesangsfrequenzen schon über die Klarheit des Sounds entscheiden können.

HF EQ

Drehen des Reglers im Uhrzeigersinn verstärkt die Frequenzen ab 12 kHz um bis zu 15 dB und gibt Becken, Gesang und elektronischen Instrumenten mehr Klarheit. Drehen entgegen gesetzt des Uhrzeigersinns reduziert Zischlaute und andere unerwünschte, hochfrequente Anteile im Signal. Wenn dieser Frequenzbereich weder verstärkt, noch abgeschwächt werden soll, muß der Regler in Mittenposition stehen.

MID EQ(HMIDund LMID)

Jeweils 2 Regler sind für die HMID und die LMID-Sektion vorhanden. Mit den oberen Reglern jeder Sektion wird die spezifische Frequenz eingestellt. In der HMID-Sektion liegt der Einstellbereich von 550 Hz bis 13 kHz und in der LMID-Sektion liegt der Einstellbereich von 50 Hz bis 1,9 kHz. Beide Bereiche können jeweils mit dem unteren Regler um 15 dB verstärkt oder abgeschwächt werden. Dieser Teil der EQ-Sektion eignet sich besonders zum Herausfiltern und Hervorheben bestimmter Klang-Eigenschaften. So liegen z.B. die Gesangsstimmen zumeist auch im Mittenbereich. Wenn Sie keinen Frequenzbereich in diesem Teil der EQ-Sektion verstärken oder abschwächen möchten, dann müssen die +/- 15 dB-Regler jeweils in Mittenposition stehen. Der Q-Faktor beträgt 1,5.

LF EQ

Drehen des Reglers im Uhrzeigersinn verstärkt die tiefen Frequenzen bis max. 60 Hz um bis zu 15 dB und eignet sich besonders für "klangliche Wärme" für Gesang oder für den zusätzlichen Druck ("Extra Punch") bei Synthesizern, Gitarren und Drums. Drehen entgegen gesetzt des Uhrzeigersinns reduziert Brummen, Bühnen-Schwingungen und verbessert dröhnenden Sound. Wenn dieser Frequenzbereich weder verstärkt, noch abgeschwächt werden soll, muß der Regler in Mittenposition stehen.

❺ INSERT

Le point d'insertion asymétrique, pré-correcteur est une rupture dans le circuit de voie, permettant d'insérer des limiteurs, des compresseurs, un correcteur spécial ou d'autres appareils de traitement de signaux. L'insertion utilise une embase jack 6.35 stéréo qui est normalement by-passée. Quand une prise est insérée, le circuit est coupé, juste avant la section CORRECTEUR. Le départ insert peut être utilisé en tant que sortie directe avant fader et avant correcteur, en créant une liaison entre l'extrémité et l'anneau du jack de sorte que le circuit ne soit pas interrompu.

❻ SORTIE DIRECTE

Les voies 1 à 16 ont une sortie directe dédiée qui permet la connexion directe aux equipements externes, par exemple magnétophones ou effets. Le niveau de ce signal avant-fader peut être écouté et visualisé sur l'afficheur bargraph en enfonçant la touche PFL de la voie appropriée.

❼ TOUCHE DIR. PRE

Les sorties directes des voies 1 à 16 sont normalement après-fader pour utilisation en départ effets ou pour régler, à l'aide du fader, le niveau d'un enregistrement. En enregistrement public, cette touche permet de passer individuellement les sorties avant-fader de manière à ce que le niveau ne soit pas affecté par les mouvements de fader durant le mixage.

❽ CORRECTEUR

Le correcteur (EQ) permet une manipulation précise du son, d'améliorer en particulier le son en sonorisation où le signal initial est souvent loin d'être idéal et où une légère accentuation ou diminution des fréquences particulières de voix peut vraiment faire une différence de clarté. Il y a trois sections permettant d'agir sur différentes plages de fréquence.

AIGUES

Tourné vers la droite, ce potentiomètre accentue de 15dB les fréquences hautes (aigu's) au-dessus de 12kHz, ajoutant de la brillance aux cymbales,aux voix et aux instruments électroniques. Tourné vers la gauche il atténue ces fréquences jusqu' à 15dB, en réduisant le souffle excessif qui peut se produire avec certains types de microphone. La position neutre est repérée par un déclic central.

HAUTS ET BAS MEDIUMS

Deux potentiomètres travaillent ensemble pour corriger les médiums. Le potentiomètre inférieur accentue ou réduit de 15dB, comme le potentiomètre du correcteur d'aigu's, mais la fréquence d'action peut être réglée par le potentiomètre supérieur sur une plage de 550Hz à 13kHz (HMID) ou de 80Hz à 1,9kHz (LMID). Ceci permet une amélioration véritablement créatrice du signal en sonorisation, cette bande médium couvrant la plage de la plupart des voix. Ecoutez soigneusement lorsque vous utilisez ces commandes pour trouver comment les caractéristiques particulières d'un signal vocal peuvent être améliorées. La position neutre est repérée par un déclic central. Note : Le coefficient Q est fixé à 1.5.

GRAVES

Tourné vers la droite, ce potentiomètre accentue de 15dB les basses fréquences (graves) au-dessous de 60Hz, en ajoutant de la chaleur aux vocaux ou du punch supplémentaire aux synthétiseurs, aux guitares et aux batteries. Tourné vers la gauche, il coupe ces fréquences de 15dB pour réduire le ronflement, ou pour éclaircir un son. La position neutre est repérée par un déclic central.

❺ INSERT POINT

L'insert point sbilanciato, pre-EQ è un'interruzione nel percorso del segnale lungo il canale, che permette a limiters, compressori, EQ speciali o altre unità di trattamento del segnale di essere aggiunte al percorso. L' Insert è un jack a 3-poli 'A' gauge che è normalmente bypassato. Il Send può essere usato come pre-fade alternativo, uscita diretta pre-EQ, se necessario, con un cavo che metta in corto circuito l'anello e la punta del jack per non interrompere il segnale.

❻ DIRECT OUT

I canali 1-16 hanno una Uscita Diretta dedicata (Direct Output) che consente il collegamento diretto ad apparecchi esterni, es. registratori o unità di effetti. Il livello dell'uscita diretta pre-fade può essere controllato premendo il tasto PFL sul relativo canale per mandare il segnale ai monitor e agli indicatori.

❼ DIR. PRE

Le uscite dirette sui canali 1-16 sono normalmente POST-FADER per l'uso con mandate di effetti o per controllare il livello di registrazione con i fader in applicazioni da studio. Per la registrazione dal vivo le uscite possono essere impostate singolarmente come PRE-FADER premendo questo tasto, così da non influenzare il livello dell'uscita diretta con il fader del mix PA principale.

❽ EQUALIZZATORE

L'Equalizzatore (EQ) permette una accurata manipolazione del suono, specialmente per migliorarlo in applicazioni PA dove il segnale originale è spesso inadeguato e leggeri aggiustamenti possono realmente fare la differenza.

HF EQ

Girando a destra questa manopola si accentuano le frequenze alte (treble) sopra 12kHz fino a 15dB, aggiungendo nitidezza ai piatti alle voci e agli strumenti elettronici. Girando a sinistra si tagliano fino a 15dB, riducendo il fruscio o i sibili che si presentano con certi microfoni. Quando non ne è richiesto l'uso lasciare la manopola fissa al centro.

MID EQ (HMID & LMID)

Ci sono due coppie di manopole che funzionano insieme per formare le sezioni HI e LO MID. Quella inferiore di ogni coppia fornisce 15dB di cut e boost, come per la regolazione HF EQ, ma la frequenza a cui questo avviene può essere impostata dalla manopola superiore su una gamma da 550Hz a 13kHz (HMID) oppure da 80Hz a 1.9kHz (LMID). Questo permette alcuni reali miglioramenti creativi del segnale in situazioni live, poichè questa banda media copre la gamma di molte voci. Ascoltare attentamente i cambiamenti durante l'uso di questi controlli così da scoprire le particolari caratteristiche e gli effetti sul segnale. Quando non ne è richiesto l'uso lasciare la manopola inferiore fissa al centro.

LF EQ

Girando la manopola a destra si aumentano le frequenza basse (60Hz e inferiori) fino a 15dB, aggiungendo calore alle voci o energia a synth, chitarre e batterie. Girando a destra si tagliano le frequenze basse fino a 15dB per ridurre il ronzio, il rumore del palco o per migliorare suoni poco chiari. Quando non ne è richiesto l'uso lasciare la manopola in posizione centrale.

❺ PUNTO DE INSERCION

El punto de inserción no balanceado, pre-EQ es un corte en el paso de la señal del canal, permitiendo añadir en éste limitadores, compresores, ecualización especial u otros procesadores de señal. La inserción es un conector jack de 1/4" de tipo A, que normalmente está en BYPASS. Cuando se inserta un jack, se corta la señal justo antes del ecualizador. El envío puede enviarse a grabación como alternativa a la salida directa pre-fader, pre-EQ si se necesita, usando un conector en el que se conectan la PUNTA y el ANILLO para no interrumpir el paso de la señal (ver a la izquierda).

❻ SALIDA DIRECTA

Los canales 1-16 disponen de una salida directa dedicada que permite la conexión directa a dispositivos externos, por ejemplo para envíos a grabadores de cinta o a unidades de efectos.

El nivel de la salida directa pre-fader puede monitorizarse pulsando el botón PFL en el canal apropiado para dirigir la señal pre-fader a los monitores y a los medidores bargraph.

❼ DIR. PRE

Las salidas directas en los canales 1-16 son normalmente POST_FADER para usarlos como envíos a efectos o para ofrecer control de fader de los niveles de grabación en aplicaciones de estudio de grabación. Para grabación de directo, las salidas pueden conmutarse individualmente a PRE_FADER pulsando este botón, de forma que el nivel de la salida directa no varíe con los ajustes del fader para la mezcla principal.

❽ ECUALIZACION

El ecualizador (EQ) permite una precisa manipulación del sonido, particularmente con el propósito de mejorar el sonido en aplicaciones de directo, donde, a menudo, la señal original difiere mucho del ideal y donde una ligera atenuación o realce sobre unas frecuencias determinadas puede realmente marcar la diferencia en cuanto a claridad.

HF EQ

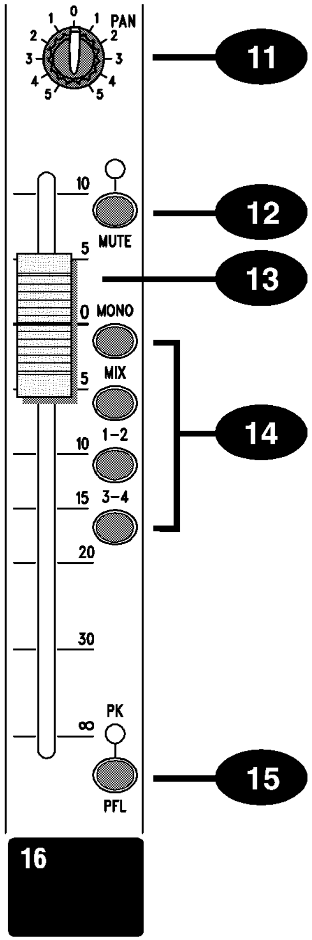
Gírelo hacia la derecha para realzar las altas frecuencias (12kHz) hasta 15 dB, añadiendo viveza a los platos, voces e instrumentos electrónicos. Gírelo hacia la izquierda para eliminar estas frecuencias hasta 15 dB, reduciendo el soplido o el excesivo énfasis de las consonantes con alto contenido en altas frecuencias, lo cual puede producirse con ciertos tipos de micrófonos. Sitúe el botón en su enclavamiento central cuando desee conseguir una respuesta plana.

MID EQ (HMID & LMID)

Se dispone de dos pares de botones que trabajan conjuntamente para formar un ecualizador de medios y agudos. El botón inferior de cada par proporciona 15 dB de realce o corte, igual que el botón HF EQ, pero la frecuencia a la que se produce esto puede ajustarse por medio del botón superior sobre una gama situada entre 550 Hz y 13 KHz (HMID) ó 80 Hz y 1.9 kHz (LMID). Esto permite una auténtica mejora creativa de la señal en situaciones de directo, ya que esta banda central cubre la gama de la mayoría de voces. Escuche cuidadosamente cuando utilice estos controles conjuntamente con el fin de encontrar la forma como puede reducirse o realzarse una característica concreta de una señal vocal. Coloque el botón inferior en su enclavamiento central cuando no lo necesite. Nota: la Q está ajustada en 1.5.

LF EQ

Gírelo hacia la derecha para realzar las bajas frecuencias (graves) hasta 15 dB a 60 Hz, añadiendo calor a las voces o "pegada" extra a los sintetizadores, guitarras y percusiones. Gírelo hacia la izquierda para atenuar las bajas frecuencias hasta 15 dB, con el fin de reducir zumbidos, vibraciones en el escenario o mejorar un sonido pobre. Coloque el botón en su enclavamiento central cuando no lo necesite.



⑨ EQ SWITCH

The EQ switch bypasses the Equalisation section when released. Alternately pressing and releasing the switch provides an easy way of comparing the equalised and unequalised signals.

⑩ AUX SENDS

These are used to set up separate mixes for FOLDBACK, EFFECTS or recording, and the combination of each Aux Send is mixed to the respective Aux Output at the rear of the mixer. For Effects it is useful for the signal to fade up and down with the fader (this is called POST-FADE), but for Foldback or Monitor feeds it is important for the send to be independent of the fader (this is called PRE-FADE). All Aux Sends are muted with the other channel outputs when the MUTE switch is pressed.

All six Aux Sends are POST-EQ, unless the EQ is bypassed using the EQ switch (see above) and are normally POST-FADE for use as effects sends or additional submixes. Aux's 1 & 2 and 3 & 4 may if required be switched in pairs to PRE-FADE by pressing the respective PRE switch, for use as foldback or monitor feeds. Aux's 5 & 6 always remain POST-FADE.

⑪ PAN

This control sets the amount of the channel signal feeding the Left and Right MIX buses, allowing you to move the source smoothly across the stereo image. When the control is turned fully right or left you are able to route the signal at unity gain to either left or right outputs individually.

⑫ MUTE

All outputs from the channel except inserts are active when the MUTE switch is released and muted when the switch is down, allowing levels to be pre-set before the signal is required.

⑬ FADER

The 100mm FADER allows precise balancing of the various source signals being mixed to the Master Section. You get most control when the input Sensitivity is set up correctly, giving full travel on the fader. See the `Setting Up & Troubleshooting' section on page 20 for help in setting a suitable signal level.

⑭ ROUTING

The channel signal may be routed to the main stereo MIX or pairs of group busses (1-2, 3-4) by pressing the respective switches, with the channel signal fed proportionately to left (1, 3) or right (2, 4) depending on the position of the PAN control (11). The channel signal may also be routed to the separate mono bus by pressing the MONO switch, unaffected by the position of the PAN control.

⑮ PFL/PEAK

When the latching PFL switch is pressed, the pre-fade, post-EQ signal is fed to the headphones, control room output and meters, where it replaces the selected monitor source. The adjacent LED lights to identify the selected channel and the PFL/AFL LED on the Master section illuminates to warn that a PFL is active. This is a useful way of listening to any required input signal without interrupting the main mix, for making adjustments or tracing problems.

When the PFL switch is released the LED serves as a PEAK indicator which illuminates approximately 4dB before clipping to give warning of a possible overload. The signal is sampled both after the HF EQ and also POST EQ.

⑨ EQ-SCHALTER

Mit dem EQ-Schalter wird die EQ-Sektion in den Signalweg geschaltet, wenn der Schalter gedrückt wird. Bei nicht gedrücktem Schalter wird die EQ-Sektion umgangen (bypass geschaltet).

⑩ AUX-SENDS

Die Ausspielwege (Aux-Sends) werden für separate Abmischungen eingesetzt, z.B. für den Musiker-Monitor, Effekte oder für die Aufnahme und die Kombination jedes Ausspielweges steht rückseitig am LX7 ein entsprechender Aux-Anschluß (6,3 mm Klinke) zur Verfügung.

Für Effekte ist es wichtig, daß der Signalpegel immer mit der Faderposition übereinstimmt. Diese Ausspielwege sind post-fade, d.h. dieses Signal ist abhängig von der jeweiligen Faderstellung (z.B. bei Effektgeräten, Hall, Echo, u.s.w.). Für das Abhören (Monitoring) ist es wichtig, daß das Signal von der Faderposition unbeeinflußt bleibt. Diese Ausspielwege sind pre-fade, d.h. das Signal wird unabhängig von der Faderstellung ausgekoppelt. Alle 6 Ausspielwege des LX7 sind post-EQ, d.h., das Signal hat die EQ-Sektion durchlaufen (siehe hierzu Abschnitt EQ-Sektion). AUX 1 und 2 und AUX 3 und 4 können paarweise von post-fade auf pre-fade umgeschaltet werden, z.B., wie oben beschrieben, für die Monitoransteuerung. AUX 5 und 6 sind dagegen immer post-fade.

⑪ PAN

Mit dem PAN-Regler wird die Stereoabbildung des linken und rechten Mix-Busses eingestellt. Ist der Regler nach links oder rechts gedreht, liegt der entsprechende Signalpegel mit einer Verstärkung =1 am linken oder rechten Ausgang an.

⑫ MUTE

Alle Kanal-Ausgänge mit Ausnahme der Inserts werden durch Drücken des MUTE-Schalters stummgeschaltet. So können z.B. Pegel schon voreingestellt werden.

⑬ FADER

Der 100 mm Fader ermöglicht ein präzises Einstellen der Kanal-Signale, die dann zur Master-Sektion geroutet werden. Die feinste Pegeleinstellung wird erreicht, wenn die Eingangsempfindlichkeit richtig voreingestellt ist (siehe hierzu Kapitel 'Richtiger Einsatz und Fehlersuche', hier sind die wichtigen Hinweise für die Signal-Einstellung gegeben).

⑭ ROUTING

Das Kanalsignal kann zum Haupt-Stereo-Mix oder paarweise (1-2, 3-4) zu den jeweiligen Gruppen-Bussen geroutet werden, in dem einer der entsprechenden Schalter gedrückt wird: MIX, 1-2, 3-4. Bei den Gruppen-Bussen wird immer das linke Signal zum ungeradzahligen Bus (1, 3) geroutet und das rechte Signal immer zum geradzahligen Bus (2,4). Die Stereoabstimmung erfolgt mit dem beschriebenen PAN-Regler. Des weiteren kann das Signal auf den Mono-Bus geroutet werden. Hierbei wird die Pegeleinstellung des PAN-Reglers nicht berücksichtigt.

⑮ PFL / PEAK

Wird der rastende PFL-Schalter gedrückt, so wird das pre-fade, post-EQ Signal zum Kopfhörer-, zum Regieraum-Monitor-Ausgang und zur LED-Anzeige geleitet und die zuvor angewählte Monitor-Quelle ersetzt. Die LED leuchtet bei gedrücktem PFL-Schalter. Des weiteren leuchtet auf der Mastersektion zusätzlich die LED PFL/AFL und zeigt an, daß ein PFL-Schalter auf einem Modul gedrückt ist. Diese pre-fade Abhörmöglichkeit ist sinnvoll, wenn ein Eingangssignal eingestellt und dabei gleichzeitig abgehört werden soll, ohne daß der Haupt-Mix unterbrochen wird. Ist der PFL-Schalter nicht gedrückt, zeigt die LED am PFL-Schalter bei ca. 4 dB unterhalb Clipping das Peak (PK) Signal an. An 2 Meßpunkten im LX7, hinter dem HF EQ und hinter der EQ-Sektion, wird das Signal abgenommen.

⑨ TOUCHE EQ

Lorsque cette touche est relâchée, la section correcteur est bypassée. Cette touche permet de comparer un signal corrigé et non corrigé.

⑩ DEPARTS AUXILIAIRES

Ceux-ci sont employés pour créer des mélanges séparés de retours musiciens, de départs EFFETS ou pour l'enregistrement. Chaque mélange est disponible à la sortie Aux à l'arrière de la console. Pour des effets, il est utile que le signal soit dépendant du fader (POST-FADE), mais pour les retours il est important que le départ soit indépendant du fader (PRE-FADE). Tous les départs Aux sont coupés lorsque la voie est coupée (touche MUTE enfoncée).

Les 6 départs Aux sont normalement APRES-CORRECTEUR (tant que la touche EQ est enfoncée, voir ci-dessus) et sont normalement POST-FADE lorsqu'ils sont utilisés comme départs effets ou mélanges supplémentaires. Les départs Aux 1 & 2 et 3 & 4 peuvent être commutés PRE-FADE si nécessaire (retour musiciens), en enfonçant la touche PRE corrspondante. Les Aux 5 & 6 restent toujours POST-FADE.

⑪ PANORAMIQUE

Ce potentiomètre dose la proportion de signal entre les côtés droit et gauche, vous permettant de déplacer le signal au sein de l'image stéréo. Quand le potentiomètre est tourné entièrement à droite ou à gauche vous pouvez diriger le signal vers les sorties droite ou gauche individuellement.

⑫ TOUCHE MUTE

Toutes les sorties de voie sauf les inserts sont actives quand la touche MUTE est relâchée et coupées quand la touche est enfoncée, permettant de prérégler le niveau avant que le signal soit activé.

⑬ FADER

Le potentiomètre rectiligne, d'une course de 100mm, permet un contrôle du niveau global de signal dans la voie. Il permet l'équilibrage précis des divers signaux de source mélangés dans le mixage final. La SENSIBILITE d'entrée doit être réglée correctement pour utiliser toute la course du potentiomètre. Voir la section réglages Initiaux à la page 9 pour le réglage correct du niveau de signal.

⑭ TOUCHES D'AFFECTATION

En appuyant sur ces touches, le signal de voie peut être dirigé vers le mixage stéréo final ou vers une paire de sorties groupes (1-2, 3-4). Selon la position du PANoramique, le signal sera dirigé proportionnellement vers la gauche (1-3) ou vers la droite (2, 4). Lorsqu'on appuie sur la touche MONO, le signal est dirigé vers une sortie mono séparée et n'est donc pas affecté par le panoramique.15

⑮ PFL

Quand la touche PFL est enfoncée, le signal PRE-FADE après-correcteur est envoyé au casque, à la sortie écoute et aux afficheurs, où il remplace la source choisie. La Led PFL adjacente s'illumine pour indentifier la voie sélectionnée et la Led PFL/AFL sur la section principale s'illumine pour avertir qu'une pré-écoute est activée. Cela permet d'écouter n'importequel signal d'entrée sans interrompre le mélange principal, pour faire des réglages ou repérer des problèmes.

Lorsque la touche PFL est relâchée, la Led sert d'indicateur de crête en s'illuminant 4dB environ avant saturation pour prévenir d'une surcharge possible. Le signal est prelevé après le correcteur d'aigu's et après la section correcteur.

⑨ TASTO EQ

Il tasto EQ bypassa la sezione equalizzazione quando non è attivo. La pressione alternata di questo tasto permette di comparare i segnali equalizzati e non equalizzati.

⑩ MANDATE AUSILIARIE (AUX SENDS)

Sono usate per regolare un insieme sonoro separato per FOLDBACK, EFFETTI o registrazione, e la combinazione di ogni Mandata Ausiliaria (Aux Send) è mixata sulla rispettiva Uscita Aux sul retro del mixer. Per gli Effetti è utile che il segnale sia sfumato con il fader (questo è chiamato POST-FADE), ma per Foldback o Monitor è importante che la mandata sia indipendente dal fader (questo è chiamato PRE-FADE). Tutte le Aux Send sono escluse quando il tasto MUTE è premuto. Tutte le sei mandate Aux sono POST-EQ, a meno che l'EQ sia bypassato con il tasto EQ, e sono normalmente POST-FADE per l'uso con mandate di effetti o submix addizionali. Se richiesto le Aux 1&2 e 3&4 possono essere impostate come PRE-FADE con il relativo tasto ed usate per foldback o mandate monitor. Le Aux 5&6 sono sempre POST-FADE.

⑪ PAN

Controlla la quantità di segnale del canale che viene mandata ai bus MIX destro e sinistro, permettendo così di muovere la sorgente all'interno dell'immagine stereo. Quando la manopola è completamente a destra o completamente a sinistra il segnale può essere indirizzato individualmente alle uscite destra o sinistra

⑫ MUTE

Tutte le uscite tranne gli inserts sono attive quando il tasto MUTE non è premuto e sono escluse in caso contrario per permettere di regolare i livelli prima che il segnale arrivi.

⑬ FADER

Il FADER da 100mm, progettato per un controllo uniforme del segnale, permette un preciso bilanciamento dei diversi segnali che vengono mixati sulla Sezione Master. Si ottiene un controllo adeguato quando la Sensibilità dell'ingresso (SENSITIVITY) è impostata correttamente, così da lasciare la massima escursione al fader. Consultare la sezione "Set Up & Guida all'individuazione dei problemi " per la corretta impostazione dei parametri

⑭ ROUTING

Il segnale di un canale può essere indirizzato al MIX stereo principale o a coppie di bus (1-2, 3-4) premendo i relativi tasti, con il segnale mandato in maniera proporzionale a destra (2,4) e sinistra (1,3) secondo la posizione del controllo PAN (11). Il segnale può anche essere indirizzato al bus mono separato premendo il tasto MONO, senza che il PAN abbia effetto in questo caso.

⑮ PFL/PEAK

Quando il tasto PFL è premuto, il segnale pre-fade, post-EQ è mandato in cuffia, all'uscita della regia e ai meter, dove sostituisce la sorgente monitor selezionata. Il LED si illumina per avvertire che un PFL è attivo. Questo è un modo utile di ascoltare ogni segnale in ingresso senza interrompere l'insieme sonoro, per fare le regolazioni necessarie o individuare problemi. Quando il tasto è rilasciato il LED funge da indicatore di picco che si illumina a circa 4 dB prima del clipping per avvertire di un possibile sovraccarico. Il segnale è campionato sia dopo l'HF EQ che POST-EQ.

⑨ CONMUTADOR DE ECUALIZACION

El conmutador de ecualización suprime el circuito de ecualización en la señal al desactivarlo. Pulsar y soltar alternativamente este botón es una manera fácil de comparar las señales ecualizada y no ecualizada.

⑩ ENVIOS AUXILIARES

Se utilizan para crear mezclas independientes para monitorización, efectos o grabación, y la combinación de todos los envíos auxiliares se mezcla sobre la correspondiente salida auxiliar, situada en la parte posterior del mezclador. Para efectos es útil que la señal suba o baje de nivel con el fader (esto se llama post-fader), pero para foldback o monitoraje es importante que el envío sea independiente del fader (esto se llama pre-fader). Todos los envíos auxiliares quedan silenciados junto a las demás salidas del canal cuando el botón MUTE es presionado.

Todos los 6 auxiliares son post-ecualización, a menos que la ecualización esté desactivada usando el interruptor EQ (ver arriba) y normalmente está en post-fader para su uso como envío de efectos o sub-mezclas adicionales. Los auxiliares 1-2 y 3-4 pueden conmutarse en pares a PRE-FADER pulsando los botones PRE respectivos para su uso como salidas de foldback o monitores. Los auxiliares 5 y 6 permanecen siempre POST-FADER.

⑪ PAN

Este control ajusta la cantidad de señal del canal que ataca los buses de mezcla izquierdo y derecho, permitiéndole mover suavemente la fuente a través de la imagen estéreo. Cuando el control se encuentra totalmente girado a la izquierda o a la derecha, podrá colocar la señal con ganancia unidad individualmente tanto en las salidas izquierda o derecha respectivamente

⑫ MUTE

Todas las salidas del canal excepto las insertadas están activas cuando el botón Mute está soltado y silenciadas cuando el botón está pulsado, dejando los niveles en pre-set, antes de que la señal sea requerida .

⑬ FADER

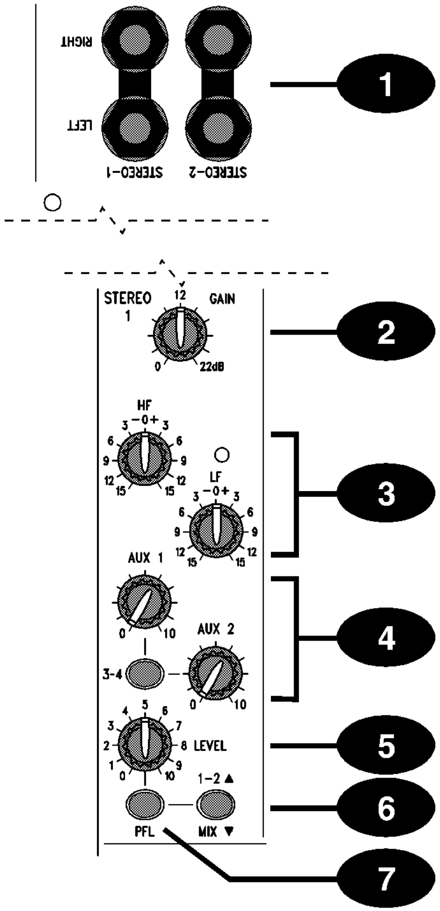
El 100mm Fader permite un balance preciso de las varias fuentes de señales para ser mezcladas en la sección Máster. Puede conseguir un mayor control cuando la entrada de sensibilidad está colocada correctamente., dejando al fader entero recorrido. Vea la sección de "Ajustes y Solución de Problemas" en la página 20 si necesita ayuda para ajustar una señal de nivel apropiada.

⑭ ROUTING

La señal del canal puede dirigirse a la mezcla estéreo principal o a pares de buses de grupo (1-2, 3-4) pulsando los conmutadores respectivos, enviándose proporcionalmente la señal a la izquierda (1, 3) o a la derecha (2, 4) dependiendo de la posición del control PAN (11). La señal del canal puede también dirigirse a bus mono separado pulsando el conmutador MONO, que no se ve afectado por el control de panorámico PAN.

⑮ PFL/PEAK

Cuando se pulsa el botón PFL, la señal pre-fader, post-EQ se envía a los auriculares, salidas de control y medidores, donde reemplaza a la fuente seleccionada de monitorización (MIX o 2TK). El LED PFL/AFL en la sección master se ilumina para avisar que un PFL está activo. Este es un modo útil de escuchar cualquier entrada sin interrumpir la mezcla principal, para hacer ajustes o hallar problemas. Al soltar el botón PFL, el LED sirve como indicador de picos, que se ilumina aproximadamente 4dB antes de la saturación para ofrecer un aviso de una posible sobrecarga. La señal se mide tras el ecualizador de HF y tras la ecualización.



STEREO INPUT CHANNEL

❶ INPUT JACKS

These high impedance inputs accept 3-pole `A' gauge (TRS) jacks. Use these inputs for sources such as keyboards, drum machines, synths, tape machines or returns from processing units. The inputs are BALANCED for low noise and top quality from professional equipment, but you can use UNBALANCED sources by wiring up the jacks as shown in the "Connecting Leads" section on page 28 in this manual, although you should then keep cable lengths as short as possible to prevent 'hum' being induced into the sound system. Mono sources may be used by plugging into the left jack only.

❷ GAIN

The GAIN control sets the input level to the channel, allowing matching to a wide range of line level sources.

❸ EQUALISER

HF EQ

Turn clockwise to boost high (treble) frequencies, adding crispness to percussion from drum machines, synths and electronic instruments. Turn anticlockwise to cut these frequencies, reducing hiss or excessive brilliance. Set the knob in the centre-detented position when not required. The control has a shelving response giving 15dB of boost or cut at 12kHz and above.

LF EQ

Turn clockwise to boost low (bass) frequencies, adding extra punch to synths, guitars and drums. Turn anticlockwise to reduce hum, boominess or improve a mushy sound. Set the knob to the centre-detented position when not required. The control has a shelving response giving 15dB of boost or cut at 60Hz and below.

❹ AUX SENDS

These are used to set up separate mixes for FOLDBACK, EFFECTS or recording, and the combination of each Aux Send is mixed to the respective Aux Output at the rear of the mixer. The sends are always PRE-FADE which is most appropriate for Foldback or Monitor feeds, but note that the Line Inputs on pairs of Mono channels may be used as alternative stereo inputs if post-fade sends are essential for effects.

❺ LEVEL

The rotary level control adjusts the overall level of the signal which is fed to the Mix or selected pair of Groups.

❻ ROUTING

The Stereo channel signal is fed either to a pair of Subgroups (switch UP) or the stereo Mix (switch DOWN), at a level set by the LEVEL control. Stereo 1 feeds to Subgroups 1 & 2, Stereo 2 feeds to Subgroups 3 & 4.

❼ PFL

When the latching PFL switch is pressed, the pre-fade, post-EQ signal is fed in mono to the headphones, control room output and meters, where it replaces the selected monitor source. The PFL/AFL LED on the Master section illuminates to warn that a PFL is active. The Left and Right meters display the PFL signal in mono.

STEREO-EINGANGSKANAL

❶ EINGANGSBUCHSEN

Diese hochohmigen Eingänge, als 6,3 mm Klinkenstecker-Buchse ausgelegt, sind für Keyboards, Drum-Machines, Synthesizer, Band-Maschinen und Returns von Effekt-Geräten vorgesehen. Die Eingänge sind symmetrisch ausgelegt, d.h., der Eingang ist rauscharm für beste Sound-Qualität bei Einsatz professionellen Equipments. Es können jedoch auch unsymmetrische Quellen angeschlossen werden. Hierzu müssen die Stecker verlötet sein, wie im Abschnitt "Anschlüsse und Stecker" gezeigt und beschrieben. Achten Sie bei unsymmetrischen Anschlüssen darauf, daß die Verbindungen möglichst kurz sind. Hierdurch wird Brummen im Sound-System verhindert. Soll an einem Stereo-Eingangs-Kanal eine Mono-Quelle angeschlossen werden, dann sollte dieser Mono-Eingang an die Buchse LEFT angeschlossen werden. Beim Anschluß LEFT durchläuft dann das Signal den linken und rechten Kanal.

❷ GAIN

Mit dem GAIN-Regler wird der Eingangspegel des Kanals eingestellt. Der Regler hat einen weiten Einstellbereich.

❸ EQUALIZER

HF EQ

Drehen des Reglers im Uhrzeigersinn verstärkt die Frequenzen ab 12 kHz um bis zu 15 dB und gibt Becken, Gesang und elektronischen Instrumenten mehr Klarheit. Drehen entgegen gesetzt des Uhrzeigersinns reduziert Zischlaute und andere unerwünschte, hochfrequente Anteile im Signal. Wenn dieser Frequenzbereich weder verstärkt, noch abgeschwächt werden soll, muß der Regler in Mittenposition stehen. Diese Klangregelung hat eine sogenannte Kuhschwanz-Charakteristik und verstärkt bzw. schwächt Frequenzen ab 12 kHz um +/- 15 dB.

LF EQ

Drehen des Reglers im Uhrzeigersinn verstärkt die tiefen Frequenzen bis max. 60 Hz um bis zu 15 dB und gibt den zusätzlichen Druck ("Extra Punch") für Synthesizer, Gitarren und Drums. Drehen entgegen gesetzt des Uhrzeigersinns reduziert Brummen und Dröhnen. Wenn dieser Frequenzbereich weder verstärkt, noch abgeschwächt werden soll, muß der Regler in Mittenposition stehen. Diese Klangregelung hat eine sogenannte Kuhschwanz-Charakteristik und verstärkt bzw. schwächt Frequenzen bei 60 Hz um +/- 15 dB.

❹ AUX-SENDS

Die Ausspielwege (Aux-Sends) werden für separate Abmischungen eingesetzt, z.B. für den Musiker-Monitor, Effekte oder für die Aufnahme und die Kombination jedes Ausspielweges steht ein entsprechender Aux-Anschluß (6,3 mm Klinke) zur Verfügung.

Die Aux Sends der Stereo-Eingangs-Kanäle sind immer pre-fade, da zum Monitoraufbau ein pre-fade Signal benötigt wird. Beachten Sie, daß die Line-Eingänge, als Mono-Kanal-Paar ausgelegt, alternativ als Stereo-Eingänge eingesetzt werden können, falls post-fade Sendewege notwendig für Effekte werden sollten.

❺ LEVEL

Mit dem LEVEL-Regler wird der Signalpegel eingestellt, der zum Mix-Bus oder zu den angewählten Gruppen-Bussen geroutet werden soll.

❻ ROUTING

Das Signal des Stereo-Kanals kann entweder zu Gruppen-Buspaaren (wenn Schalterposition oben) oder zum Stereo-Mix-Bus (wenn Schalter gedrückt) geroutet werden. Der Signalpegel wird mit dem LEVEL-Regler eingestellt. Stereo 1 steuert die Subgruppen 1 und 2 an, Stereo 2 steuert Subgruppen 3 und 4 an.

VOIE D'ENTREE STEREO

1 EMBASES JACK

Ces entrées sur embase jack STEREO sont destinées à des sources telles que des claviers, bo"tes à rythmes, synthétiseurs, magnétophones ou effets. Les entrées sont SYMETRIQUES pour un meilleur rapport signal/bruit et pour du matériel professionnel, mais vous pouvez utiliser des sources ASYMETRIQUES en câblant comme montré à la section «Câblage» page 28 de ce manuel ; dans ce cas, gardez des câbles aussi courts que possible. Des sources mono peuvent être employées en branchant la prise gauche seulement.

2 GAIN

Le potentiomètre de GAIN règle la sensibilité d'entrée de la voie, permettant de s'adapter à une large gamme de sources à niveau ligne.

3 CORRECTEUR

AIGUES

Tourné vers la droite, ce potentiomètre accentue de 15dB les fréquences hautes (aigu's) à 12kHz, ajoutant de la brillance aux percussions, boîtes à rythmes, synthétiseurs et aux instruments électroniques. Tourné vers la gauche il atténue les fréquences en réduisant le souffle excessif. La position neutre est répérée par un déclic central.

GRAVES

Tourné vers la droite, ce potentiomètre accentue de 15dB les basses fréquences (graves) à 60 Hz en ajoutant du punch supplémentaire aux synthétiseurs, guitares et batteries. Tourné vers la gauche, il coupe les basses fréquences pour réduire le ronflement, la résonance ou pour éclaircir un son. La position neutre est répérée par un déclic central.

4 DEPARTS AUX

Ceux-ci sont employés pour créer des mélanges séparés de retours musiciens, de départs effets ou l'enregistrement. Chaque mélange est disponible à la sortie Aux à l'arrière de la console. Les départs sont toujours indépendants du fader (PRE-FADE) ce qui convient mieux pour les retours musiciens. Notez que les entrées ligne des voies mono peuvent être utilisées comme entrée stéréo si les départs doivent être dépendants du fader (POST-FADE).

5 LEVEL

Ce potentiomètre rotatif règle le niveau général du signal envoyé vers le mix ou vers une paire de groupes.

6 TOUCHE D'AFFECTATION

En appuyant sur cette touche, le signal de voie est dirigé vers le mixage stéréo. Lorsque la touche est relâchée, le signal est envoyé vers une paire de sous-groupes. Le niveau est réglé par le potentiomètre LEVEL. Le signal de la voie stéréo 1 est dirigé vers les sous-groupes 1 & 2, celui de la voie stéréo 2 vers les sous-groupes 3 & 4.

7 PFL

Quand la touche PFL est enfoncée, le signal PRE-FADE et après correcteur est dirigé en mono vers le casque, vers la sortie écoute et vers les afficheurs, où il remplace la source choisie. La Led PFL/AFL sur la section Master s'illumine pour avertir qu'une pré-écoute est active. Les afficheurs gauche et droit affichent le signal PFL mono.

INGRESSO STEREO

1 INGRESSI JACK

Questi ingressi ad alta impedenza accettano jack a 3-poli `A' gauge (TRS), Questi ingressi vanno usati per sorgenti come tastiere, drum machines, synth, registratori o come ritorno dalle unità di "processing". Gli ingressi sono BILANCIATI per diminuire il rumore e avere la massima la qualità dalle apparecchiature professionali, ma è possibile usare sorgenti NON BILANCIATE collegando i jack come mostrato nella relativa sezione del manuale (pag. 28), sebbene dobbiate mantenere la lunghezza dei cavi il più corto possibile per ridurre le interferenze. Le sorgenti mono possono essere inserite su entrambi i percorsi utilizzando solo il jack di sinistra.

2 GAIN

Il controllo GAIN regola il livello di ingresso al canale, permettendo la modulazione di diverse sorgenti line.

3 EQUALIZZATORE

HF EQ

Girando la manopola in senso orario le alte (treble) frequenze aumentano aggiungendo brillantezza a percussioni, synth e strumenti elettronici. La rotazione in senso antiorario le diminuisce riducendo il fruscio o l'eccessiva brillantezza. Quando non ne è richiesto l'uso posizionare il controllo al centro. Ha una risposta di (15dB a 12kHz e oltre.

LF EQ

Girando la manopola in senso orario le basse (bass) frequenze (60kHz e inferiori) aumentano, aggiungendo incisività a sintetizzatori, chitarre e batteria. La rotazione in senso antiorario le diminuisce, riducendo il ronzio, il rumore del palco o migliorando un suono non chiaro. Quando non ne è richiesto l'uso posizionare il controllo al centro. Ha una risposta di (15dB a 60kHz e inferiori.

4 MANDATE AUSILIARIE (AUX SEND)

Sono usate per regolare un insieme sonoro separato per FOLDBACK, EFFETTI o registrazione, e la combinazione di ogni Mandata Ausiliaria (Aux Send) è mixata sulla rispettiva Uscita Aux sul retro del mixer. Le mandate sono sempre PRE-FADE, che l'impostazione adatta per foldback o mandate monitor, ma gli ingressi Line a coppie di canali mono possono essere usate come ingressi stereo alternativi se gli per effetti è essenziale che le mandate siano post-fade.

5 LEVEL

Questa manopola regola il livello generale del segnale mandato al Mix o alla coppia di Gruppi selezionata.

6 ROUTING

Il segnale del canale stereo viene mandato ad una coppia di sub-gruppi (interruttore rilasciato) o al Mix stereo (interruttore premuto) al livello impostato con il relativo controllo. Stereo 1 alimenta i sub-gruppi 1&2, mentre Stereo 2 alimenta i sub-gruppi 3&4.

7 PFL

Quando il tasto PFL è premuto, il segnale pre-fade, post-EQ è mandato in cuffia, all'uscita della regia e ai meter, dove sostituisce la sorgente monitor selezionata. Il LED PFL/AFL nella Sezione Master si illumina per avvertire che un PFL è attivo. I meter destro e sinistro mostrano il segnale PFL mono.

CANAL DE ENTRADA ESTEREO

1 JACKS DE ENTRADA

Estas entradas de alta impedancia aceptan jacks de 1/4" (TRS) de 3. Utilice estas entradas para fuentes tales como teclados, cajas de ritmo, sintetizadores, magnetófonos o como retornos de procesadores de efectos. La entrada es BALANCEADA y de alta calidad para equipos profesionales, pero sin embargo, puede utilizar fuentes NO BALANCEADAS cableando los conectores tal como se muestra en la sección "Conectando lo cables" de la página 28 de este manual, aunque deberá mantener la longitud de los cables tan corta como le sea posible para prevenir ruidos indeseados. Las fuentes mono pueden alimentar ambos lados si se conectan sólo en el jack izquierdo.

2 GANANCIA

El control de ganancia ajusta el nivel de entrada al canal, permitiendo su adaptación a una amplia gama de fuentes de nivel de línea.

3 ECUALIZACION

HF EQ

Gírelo hacia la derecha para realzar las altas frecuencias (agudos), añadiendo viveza a los sonidos de percusión procedentes de una caja de ritmos, sintetizadores e instrumentos electrónicos. Gírelo hacia la izquierda para atenuar estas frecuencias, reduciendo el soplido o el brillo excesivo. Coloque el botón en su enclavamiento central cuando no lo utilice. El control posee una respuesta del tipo shelving, proporcionando 15 dB de realce o atenuación a la frecuencia fija de 12 Khz.

LF EQ

Gírelo hacia la derecha para realzar las frecuencias bajas (graves), añadiendo mayor "pegada" a sintetizadores, guitarras y percusiones. Gírelo hacia la izquierda para reducir zumbidos, sonidos atronadores o para mejorar un sonido poco enérgico. Coloque el botón en su enclavamiento central cuando no lo utilice. El control posee una respuesta de tipo shelving, proporcionando 15 dB de realce o atenuación a la frecuencia fija de 60 Hz.

4 ENVIOS AUXILIARES

Se utilizan para crear mezclas independientes para monitorización, efectos o grabación, y la combinación de todos los envíos auxiliares se mezcla sobre la correspondiente salida auxiliar, situada en la parte posterior del mezclador.

Los envíos son siempre PRE-FADER, que es más apropiado para salidas de foldback o monitores, pero observe que las entradas de línea en pares de canales mono pueden usarse como entradas estéreo alternativas si se requieren envíos post-fader para efectos.

5 NIVEL

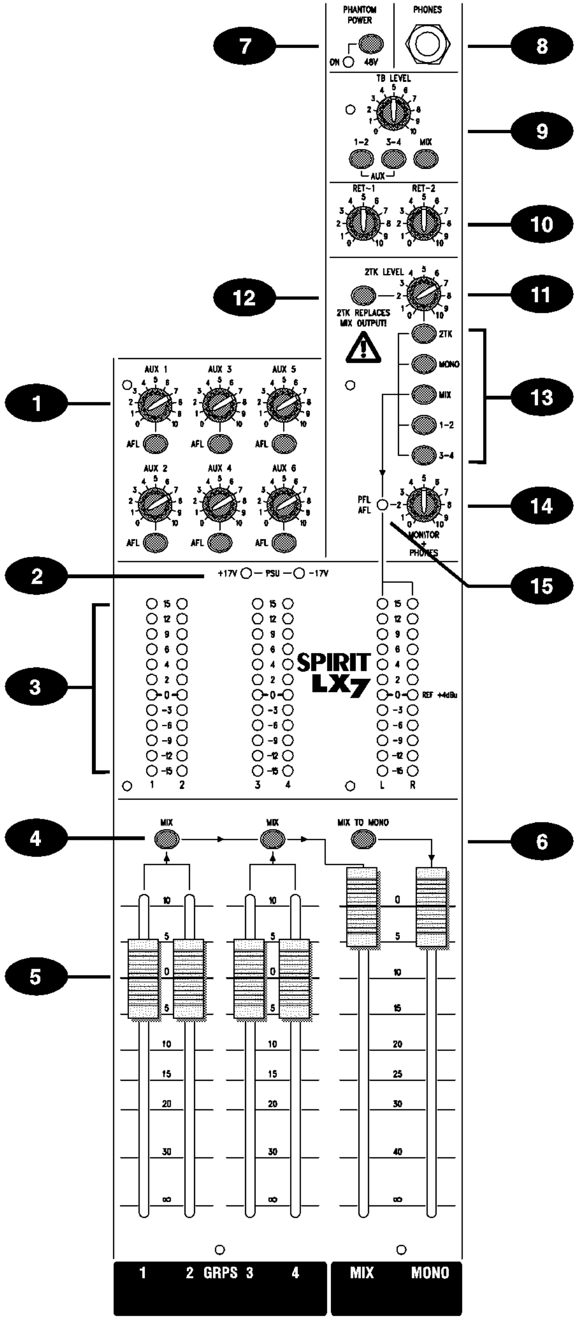
El control rotativo de nivel ajusta el nivel general de la señal que se envía a la mezcla o al par de grupos seleccionados.

6 ROUTING

La señal del canal estéreo se envía a un par de subgrupos (botón arriba) o a la mezcla estéreo (botón abajo), a un nivel ajustado por el control de NIVEL. El estéreo 1 se envía a los subgrupos 1 y 2, y el estéreo 2 a los subgrupos 3 y 4.

7 PFL

Cuando se pulsa el botón PFL, la señal pre-fader, post-EQ se envía a los auriculares, salidas de control y medidores, donde reemplaza a la fuente seleccionada de monitorización (MIX o 2TK). El LED PFL/AFL en la sección master se ilumina para avisar que un PFL está activo. Los medidores izquierdo y derecho muestran la señal PFL en mono.



MASTER SECTION

❶ AUX MASTERS

Each of the six AUX outputs has a master output level control and associated AFL switch.

AUX AFLs

Just as the Channel PFL switches allow *pre-fade listening*, so you can monitor each AUX output *after the level control* by pressing the AFL switch. This routes the AUX output signal to the MONITOR or PHONES, replacing any existing signal which is selected. The METERS also switch from the selected source to display the PFL/AFL signal and the PFL/AFL LED lights to warn that a PFL or AFL switch is pressed. When you release the switch the Monitor swaps back to the previous source.

❷ POWER INDICATORS

These LEDs light to show that power is connected to the console and that the internal power supply is operating correctly.

❸ BARGRAPH METERS

3-colour peak reading BARGRAPH METERS are provided to monitor the four Subgroup outputs and the selected Monitor + Phones source (2TK, Mono, Mix or Groups), giving you a constant warning of excessive peaks in the signal which might cause overloading. Aim to keep the signal within the amber segments at peak levels for best performance.

Similarly, if the output level is too low and hardly registering at all on the meters, the level of background noise may become significant. Take care to set up the input levels for best performance.

When any PFL or AFL switch is pressed, the L & R meters automatically switch to show the selected PFL/AFL signal on both meters, in mono.

❹ MIX

Pressing the Mix switch routes the post-fade Subgroup signals in pairs to the main Mix. Groups 1 & 3 are routed to Mix L, Groups 2 & 4 are routed to Mix R.

❺ MASTER FADERS

The MASTER FADERS set the final level of the Subgroup and Mix L & R outputs. These should normally be set close to the `0' mark if the input GAIN settings have been correctly set, to give maximum travel on the faders for smoothest control.

❻ MIX TO MONO

Pressing this switch routes the post-fade Mix L/R outputs to the Mono bus to create a separate mono mix to feed, for example, an induction loop or centre cluster. Note: If there are input channels which are routed both to Mix and Mono, pressing this switch will have an additive effect which may lead to feedback.

❼ 48V (Phantom Power)

⚠ Many professional condenser mics need an external powering voltage, normally 48V, also known as PHANTOM POWER. This is a method of sending a powering voltage down the same wires as the mic signal. Press the switch to supply the 48V power to all of the MIC inputs. The adjacent LED illuminates when the power is active.

❼ PFL

Wird der rastende PFL-Schalter gedrückt, wird das pre-fade, post-EQ Signal zum Kopfhörer-Signal, zum Regieraum-Monitor-Ausgang und zur LED-Anzeige geleitet, wobei das zuvor anliegende Monitor-Signalüberschrieben wird. Des weiteren leuchtet auf der Mastersektion zusätzlich die LED PFL/AFL und zeigt an, daß ein PFL-Schalter auf einem Kanal gedrückt ist. Linke und rechte LED-Kette zeigen ein Mono-Signal an.

MASTER SEKTION

❶ AUX MASTERS

Jeder der 6 AUX-Ausgänge hat einen eigenen Haupt-Ausgangs-Pegelregler und einen zugehörigen AFL-Schalter (After-fade = Fadergeregeltes Signal).

AUX-AFL

Genauso, wie die einzelnen PFL-Schalter das pre-fade Hören des Signals ermöglichen, so kann jeder Aux-Ausgang nach dem Pegelregler abgehört werden. Hierzu muß der AFL-Schalter gedrückt werden. Dadurch wird das Aux-Ausgangssignal auf die Monitor- oder Phones-Ausgänge geschaltet. Die LED-Anzeige wird zusätzlich zur Anzeige des PFL/AFL-Signals umgeschaltet. Die LED PFL/AFL leuchtet und zeigt wiederum an, daß ein PFL- oder AFL-Schalter gedrückt ist. Wird der Schalter wieder gedrückt, so wird auf die vorher angewählte Quelle wieder zurückgeschaltet.

❷ POWER-ANZEIGEN

Die LEDs leuchten, wenn das interne Netzgerät des LX7 ordnungsgemäß arbeitet.

❸ LED-AUSSTEUERUNGSANZEIGE

Sechs 3-farbige 12-Segment-LED-Ketten dienen als Aussteuerungsanzeigen. Angezeigt werden auf den LED-Ketten die Subgruppen-Ausgangssignale und die angewählte Monitor- und Phones-Quelle (2TK, Mono, Mix oder Gruppen). So haben Sie immer einen Überblick über richtige Signalpegel-Einstellungen. Die roten LEDs zeigen beim Aufleuchten den PEAK-Bereich an. Beachten Sie also, daß die amberfarbenen LEDs immer leuchten. Dann können Sie davon ausgehen, daß die Signalpegel richtig eingestellt sind. Ein zu niedrig eingestelltes Signal könnte ein zu hohes Hintergrundrauschen verursachen. Ist einer der PFL- oder AFL-Schalter gedrückt, wird das entsprechende PFL- oder AFL-Signal auf der Aussteuerungsanzeige in Mono angezeigt.

❹ MIX

Durch Drücken des MIX-Schalters werden die post-fade Subgruppensignale paarweise dem Haupt-Mix zugewiesen. Die Gruppen 1 und 3 werden zum Mix L, die Gruppen 2 und 4 werden zum Mix R geroutet.

❺ MASTER FADER

Mit den Master Fadern wird der endgültige Pegel der Subgruppen und der Mix L und R Ausgänge eingestellt. Normalerweise sollten diese Fader bei korrekter Kanal-Eingangs-Vorverstärkung auf '0' stehen, um eine maximale Faderauflösung für eine lineare Einstellung zu gewähren.

❻ MIX TO MONO

Mit diesem Schalter werden die post-fade Mix L/R Ausgänge zum Mono-Bus geroutet. Hierdurch kann ein separater Mono-Mix erzeugt werden, mit dem beispielsweise ein Center Cluster angesteuert wird.

HINWEIS: Beachten Sie, daß bei Eingangskanälen, die dem Mix- und Mono-Bus zugewiesen sind, Rückkopplungen entstehen, die zu Beschädigungen führen können.

SECTION DE SORTIE (MASTER)

❶ GENERAUX D'AUXILIAIRES

Chacune des six sorties Aux a un réglage de niveau général et une touche associée AFL.

AFL AUXILIAIRES

Comme les touches PFL sur les voies, vous pouvez contrôler chaque sortie AUX en appuyant sur la touche AFL. Ceci dirige le signal AUX après contrôle du niveau, vers l'écoute ou le casque, se substituant au signal existant. Les AFFICHEURS se libèrent également de la source choisie pour afficher le signal de PFL/AFL et la Led PFL/AFL s'illumine pour avertir qu'une touche de PFL ou d'AFL est appuyée. Quand vous relâchez la touche, la source initiale est restituée.

❷ INDICATEUR DE MISE SOUS TENSION

Ces Leds s'illuminent pour indiquer que la console est sous tension.

❸ AFFICHEURS BARGRAPH

Les afficheurs Bargraph tricolores suivent la sélection Monitor et casque pour afficher le niveau des quatre sorties sous-groupes, du 2-PISTES, de la source mono, du Mix ou des Groupes, vous donnant un avertissement constant des crêtes excessives dans le signal qui pourraient causer une surcharge. Veillez à maintenir le signal à la limite des segments jaunes au niveau maximal. De même, si le niveau de sortie est trop bas et à peine visible sur les afficheurs, le niveau du bruit de fond peut devenir significatif. Prenez soin de régler les niveaux d'entrée au mieux.

Quand une touche de PFL ou AFL est appuyée, les afficheurs visualisent le signal de PFL/AFL choisi sur les deux afficheurs, en mono.

❹ TOUCHE MIX

Appuyez sur cette touche pour diriger par paire les signaux post-fade des sous-groupes vers le mixage final. Les groupes 1 & 3 sont dirigés vers le mix gauche, les groupes 2 & 4 sont dirigés vers le mix droit.

❺ GENERAUX

Les généraux réglent le niveau final du mélange, avec des tirettes séparées pour chaque sortie. Celles-ci doivent normalement être placées près de la marque 0 si le GAIN d'entrée a été correctement réglé, pour bénéficier au mieux de la course de la tirette.

❻ TOUCHE MIX TO MONO

Appuyez sur cette touche pour envoyer une somme mono post-fade des signaux des Mix gauche et droit. Cela permet de créer un mélange mono séparé pour attaquer par exemple une boucle d'induction ou un cluster central. Note : Si l'on appuie sur cette touche alors que des voies d'entrée sont dirigées à la fois vers les sorties Mix et Mono on risque de provoquer un Larsen.

❼ Alimentation fantôme 48V ⚠

Beaucoup de micros professionnels statiques ont besoin d'une alimentation FANTOME, qui applique une tension d'alimentation sur les conducteurs du câble micro. Appuyer sur la touche +48V pour mettre en service l'alimentation sur toutes les entrées MICRO. La Led adjacente s'illumine quand l'alimentation est activée.

SEZIONE MASTER

❶ AUX MASTER

Ciascuna delle sei uscite AUX ha un controllo master del livello di uscita ed un tasto AFL.

AUX AFL

Come i tasti PFL permettono l'ascolto pre-fade, così è possibile monitorare ogni uscita AUX dopo che ne è stato regolato il livello usando il tasto AFL. Questo indirizza il segnale d'uscita su MONITOR o PHONES (cuffie), prendendo il posto di ogni segnale presente. Anche i METERS commutano dalla sorgente selezionata per mostrare il segnale PFL/AFL ed il relativo LED si illumina per avvertire che uno dei due tasti è premuto. Quando viene rilasciato il tasto il Monitor torna alla sorgente precedente.

❷ INDICATORI DI TENSIONE

Questi LED si illuminano per mostrare che la consolle è alimentata e che l'alimentatore interno funziona correttamente.

❸ INDICATORI (METER) A BARRE

Gli indicatori a barre a tre colori controllano i quattro sub-gruppi e la sorgente Monitor + Cuffie (2TK, Mono, Mix o Gruppi) avvertendo in caso di picchi eccessivi che potrebbero causare sovraccarico. Tentare di tenere il segnale nella sezione arancione per un miglior rendimento.

Allo stesso modo, se il livello di uscita è troppo basso e viene rilevato faticosamente dai meter, il livello del rumore di fondo potrebbe divenire eccessivo. Fare attenzione ai livelli di ingresso e regolarli con cura.

Quando uno dei tasti **PFL** o AFL è premuto, i meter commutano per mostrare il segnale PFL/AFL su entrambe le file, in modalità mono.

❹ MIX

Premendo questo tasto i segnali post-fade dei sub-gruppi vengono indirizzati a coppie al Mix principale. I gruppi 1&3 vanno al MIX L (sinistro) mentre i gruppi 2&4 vanno al MIX R (destro).

❺ MASTER FADERS

I MASTER FADERS regolano il livello finale delle uscite Sub-Group e Mix destra e sinistra. Normalmente dovrebbero essere vicini a "0" se il GAIN d'ingresso è stato impostato correttamente, così da avere la massima escursione sui fader per un controllo più uniforme.

❻ MIX TO MONO

Questo switch pilota le uscite Mix post-fade destra e sinistra sul bus Mono, per creare un mix mono separato da mandare, ad esempio, ad un cluster centrale. Nota Se ci sono ingressi indirizzati a Mix e Mono, premendo questo tasto si crea un effetto che può portare al feedback.

❼ ALIMENTAZIONE PHANTOM 48V ⚠

Molti microfoni a condensatore professionali hanno bisogno di alimentazione esterna, solitamente a 48V, nota come alimentazione Phantom. Questo è un metodo per mandare il segnale di alimentazione sullo stesso cavo del segnale microfonico. Per alimentare a 48V tutti gli ingressi microfonici premere il tasto relativo. Il led si illumina per indicare l'attivazione.

SECCION MASTER

❶ MASTERS DE AUXILIARES

Cada una de las seis salidas de auxiliar disponen de un control de nivel master y un interruptor AFL asociado.

AFL DE AUXILIAR

Igual que los interruptores PFL de los canales, puede monitorizar cada salida de auxiliar pulsando el botón AFL. Esto envía la señal de salida de auxiliar al monitor o a los auriculares, reemplazando cualquier señal existente (normalmente el monitor recibe MIX, SUB o 2TK, ver más arriba). Los medidores también cambian de la fuente seleccionada a la señal PFL/AFL, y el LED PFL/AFL se ilumina para indicar que hay un botón PFL/AFL pulsado. Al soltar el botón, el monitor vuelve a la fuente anterior.

❷ INDICADOR DE ALIMENTACION

Este LED se ilumina para mostrar que la consola está alimentación está conectada y que la fuente de alimentación interna opera correctamente.

❸ MEDIDORES BARGRAPH

Los medidores de picos bargraph con 3 colores monitorizan las cuatro salidas de Subgrupo y la fuente seleccionada de Monitor + Auriculares (2TK, Mono, Mix o Grupos), ofreciéndole un aviso constante de los picos excesivos en la señal que pueden causar saturación. Procure mantener la señal dentro de los segmentos ámbar a niveles de pico para obtener el mejor comportamiento.

Si el nivel de salida es demasiado bajo, y casi no se ve en los medidores, el nivel de ruido de fondo puede ser significante. Preocúpese de ajustar los niveles de entrada para una mejor respuesta.

Al pulsar cualquier botón PFL o AFL, los medidores pasan a mostrar la señal seleccionada en ambos medidores, en mono.

❹ MIX

Pulsando el botón MIX se dirige las señales de los subgrupos estéreo en pares a la mezcla principal. Los grupos 1 y 3 se dirigen a la mezcla izquierda, y los grupos 2 y 4 se envían a la mezcla derecha.

❺ FADERS MASTER

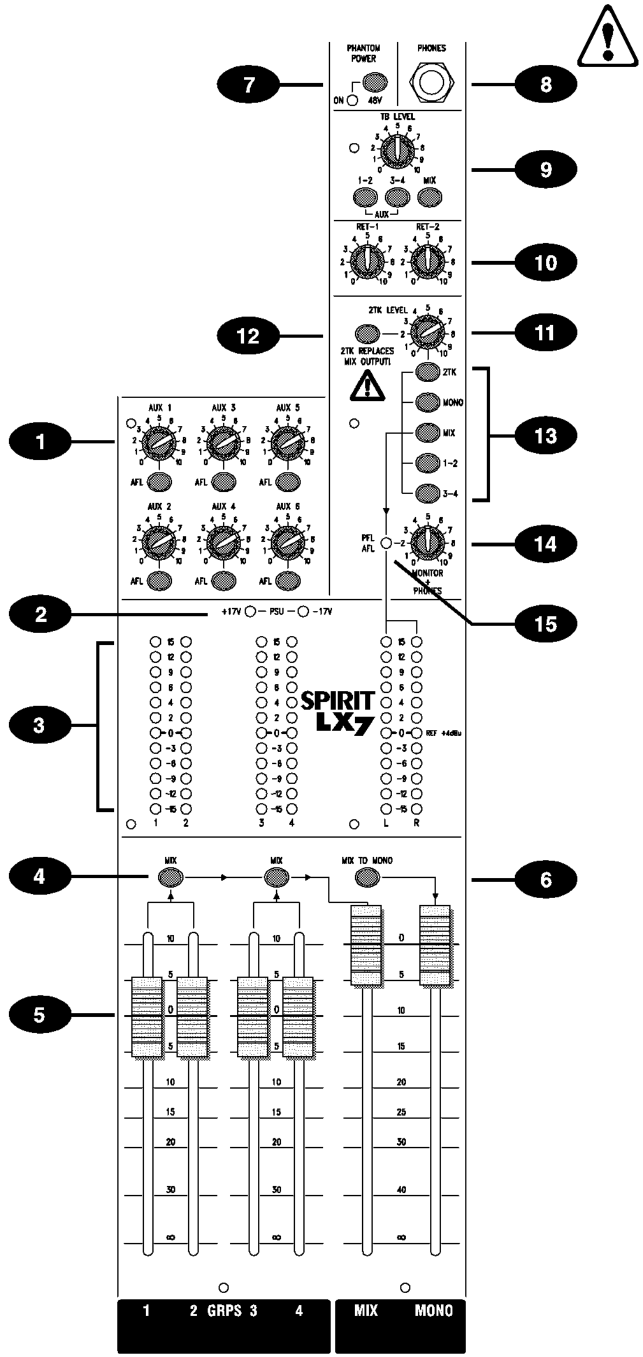
Los faders master ajustan el nivel final de las salidas de mezcla izquierda y derecha. Normalmente deberán estar cerca de la posición "0" si los ajustes de ganancia se han efectuado correctamente, para ofrecer el recorrido máximo en los faders para un control más fino.

❻ MIX TO MONO

Pulsando este botón se dirigen las salidas post-fader de mezcla L y R al bus mono para crear una mezcla mono separada para, por ejemplo, un bucle de inducción o un cluster central. Nota: Si hay canales de entrada dirigidos a MIX y a MONO, al pulsar este botón se producirá un efecto de suma que puede producir realimentaciones.

❼ 48V (Alimentación Phantom) ⚠

Muchos micrófonos profesionales de condensador requieren una alimentación externa, normalmente de 48V, conocida también como ALIMENTACIÓN PHANTOM. Es un método de enviar la alimentación a través de los mismos cables que la señal del micrófono. Pulse este botón para ofrecer la alimentación de 48V a todas las entradas de micrófono. El LED adyacente se ilumina cuando se activa la alimentación.



DO NOT USE unbalanced mics with 48V switched on as they may be damaged by the phantom power voltage. Balanced dynamic mics and leads can normally be used with 48V switched on (contact your microphone manufacturer for guidance)

Mics should always be plugged in, and all output faders set to minimum before switching 48V ON to avoid damage to external equipment

8 PHONES

The PHONES output appears on a 3-pole 1/4" jack, suitable for headphones with an impedance of 200Ω or higher.

9 TB LEVEL

A balanced input is provided for a Talkback microphone. The signal may be routed selectively to Aux 1/2 or 3/4 (which might typically be used for performers' foldback) or Mix L/R by pressing the appropriate switches. The signal level is adjusted by the TB LEVEL control.

10 STEREO RETURNS

Two balanced Stereo Returns are available for the outputs of effects units and are mixed directly to the Mix L/R busses at a level set by the RET-1 or RET-2 controls. If a mono source is used, plugging into the Left jack only automatically feeds to the signal to both Left and Right.

11 2TK LEVEL

The rotary control sets the level of the 2 Track Tape input, which is routed to the headphones, monitor outputs and meters, or directly to the Mix outputs by pressing the adjacent switch (12). These unbalanced inputs, on RCA phono connectors, are ideal to connect the playback of a tape machine for monitoring.

12 2TK REPLACES MIX OUTPUT!

This switch does what it says! When pressed the Mix output is switched to the 2 Track input and is an ideal way of feeding pre-show music to the main outputs in live applications without using up valuable input channels. For example, with the switch pressed and a CD connected and playing pre-show music, the engineer can be setting up channel levels, adjusting EQ, talking back to performers and monitoring any part of the mix (including final Mix) without affecting the sound to the audience. Releasing the switch instantly swaps the Mix Outputs back to the mixer and cuts off the CD.

CAUTION: Pressing this switch cuts off the normal Mix L/R signal and it should therefore NOT be used during live performance or recording

13 MONITOR SOURCE SELECT

These switches allow a choice of 2TK, Mono, Mix or Groups as the source for the Phones, Monitor outputs and meters, and may be selected individually or as a combination. **NOTE:** If NO switches are pressed, there will be NO signal on the meters or monitors!

7 48V (Phantom-Spannung) ⚠

Viele professionelle Kondensator-Mikrofone benötigen eine externe Spannungs-Versorgung, die sogenannte Phantom-Spannung (48V). Mit dieser Methode wird die Spannung auf die gleichen Leitung gelegt, auf denen auch das Mikrofon-Signal anliegt. Durch Drücken des Schalters wird die Phantom-Spannung auf alle MIC XLR-Eingänge geschaltet. Die zugehörige LED leuchtet, wenn die Phantom Spannung geschaltet wird.

Schließen Sie niemals unsymmetrische Mikrofone an, wenn die Phantom-Spannung (48V) eingeschaltet ist. Hier besteht Gefahr einer Beschädigung im Mischpult!

Dynamische Mikrofone und Leitungen können normalerweise mit Phantom-Spannung betrieben werden. Wir empfehlen Ihnen, beachten Sie die Bedienungsanleitung Ihres Mikrofons oder informieren Sie sich beim entsprechenden Hersteller.

Alle erforderlichen Mikrofone sollten eingesteckt und die Ausgangsfader heruntergepegelt sein, bevor der 48 V-Schalter betätigt wird. Hierdurch werden Beschädigungen am externen Equipment verhindert.

8 PHONES

Der Kopfhörer-Anschluß ist als 6,3 mm Klinkenstecker-Buchse ausgelegt. Die Eingangsimpedanz dieses Anschlusses beträgt 200 Ω oder höher.

9 TB LEVEL

Anweisungen für die Musiker können über die Aux 1,2 oder Aux 3,4 oder über die Mix L/R Ausgänge erteilt werden, wenn ein sogenanntes Talkback-Mikrofon angeschlossen wird (hierzu muß der entsprechende Schalter gedrückt sein). Der Anschluß ist symmetrisch ausgelegt. Der Kommando-Signal-Pegel wird mit dem TB LEVEL-Regler eingestellt.

10 STEREO RETURNS

Zwei symmetrische Stereo Returns stehen für den Anschluß von Effekt-Geräten zur Verfügung und die Return-Signale können direkt auf die Mix L/R-Busse gemischt werden. Der Pegel wird mit dem RET-1 oder dem RET-2 Regler eingestellt. Wird nur ein Mono-Gerät eingeschleift, nutzen Sie wieder die linke Buchse, da hier das Signal beiden Kanälen zugewiesen wird.

11 2TK LEVEL

Mit diesem Dreh-Regler wird der Pegel der 2-Spur Bandmaschinen-Eingänge eingestellt, der zum Kopfhörer-Ausgang, zu den Monitor-Ausgängen oder direkt zu den Mix-Ausgängen geroutet wird. Hierbei muß der jeweilige Schalter (12) gedrückt sein. Die unsymmetrischen Cinch-Anschlüsse dienen zum Anschluß eines Tonbandgeräts, um die Wiedergabe abhören zu können.

12 2TK REPLACES MIX OUTPUT (2TK ersetzt den Mix-Ausgang)

Dieser Schalter veranlaßt genau das, was seine Bezeichnung sagt. Wird der Schalter gedrückt, werden die 2-Spur Bandmaschinen-Eingänge auf die Mix-Ausgänge geschaltet. So können beispielsweise bei Live-Veranstaltungen Zwischeneinspielungen über die Haupt-Ausgänge erfolgen, ohne das hierfür Eingangskanäle verwendet werden müssen. Der große Vorteil dieser Einrichtung wird am folgenden Beispiel ersichtlich: Während ein angeschlossener CD-Spieler für die Zwischeneinspielung sorgt, kann der Toningenieur alle Kanal-Pegel einstellen, EQs abstimmen, Rücksprache zu den Musikern halten und jeden Teil des Mixes (inklusive des Haupt-Mixes) zur Kontrolle abhören, ohne daß der eingespielte Sound zum Publikum in irgend einer Weise gestört oder unterbrochen wird. Wird der Schalter erneut gedrückt, wird der Sound des Mix-Ausganges wieder übertragen und die 2TK-Einspielung, hier die CD, wird unterbrochen.

ATTENTION : les micros asymétriques peuvent être endommagés par la tension présente. Des micros dynamiques symétriques peuvent normalement être utilisés avec l'alimentation fantôme (contacter votre constructeur de microphone pour des conseils).

Les micros doivent toujours être branchés, et tous les faders de sortie réglées au minimum avant de mettre en service l'alimentation fantôme pour éviter d'endommager le matériel externe.

⑧ *EMBASE CASQUE*

Cette embase jack stéréo 6.35 convient pour des casques dont l'impédance est supérieure ou égale à 200Ω

⑨ *NIVEAU D'ORDRES (TB LEVEL)*

Une entrée symétrique est prévue pour un microphone d'ordres. Le signal peut être dirigé vers les Auxiliaires 1/2 ou 3/4 (destinés au retour musiciens) ou vers le mix gauche/droit en appuyant sur la touche correspondante. Le niveau est réglé par le potentiomètre TB LEVEL.

⑩ *RETOURS STEREO*

Deux retours stéréo symétriques sont disponibles pour les effets qui sont dirigés directement vers les sorties mix gauche/droite. Le niveau est réglé par les potentiomètres RET-1 et RET-2. Si la source est mono, on utilise l'embase jack gauche et le signal est dirigé automatiquement vers la droite et la gauche.

⑪ *NIVEAU 2-PISTES*

Le potentiomètre rotatif règle le niveau du retour 2-pistes dont le signal est dirigé vers le casque, les enceintes et l'afficheur ou directement vers les sorties Mix si l'on appuie sur la touche correspondante (12). Le retour 2 pistes, idéal pour écouter un magnétophone à bande, s'effectue sur des connecteurs RCA/Cinch asymétriques.

⑫ *TOUCHE «2TK REPLACES MIX OUTPUT !»*

«Le 2-pistes remplace la sortie mix» : cette touche fait ce qu'elle dit ! Appuyez sur cette touche pour injecter le retour 2 pistes dans le Mix stéréo. C'est un moyen idéal en sonorisation pour diffuser de la musique d'attente sans utiliser une voie d'entrée. Par exemple, lorsque la touche est enfoncée et qu'un lecteur CD diffuse de la musique d'attente, l'ingénieur du son peut régler le niveau des voies, le correcteur, parler aux musiciens et écouter toute partie du mix (même le mixage final) sans affecter le son. Dès que l'on relâche cette touche, le lecteur CD est coupé et les sorties mix restituées.

ATTENTION : lorsqu'on appuie sur cette touche le signal mix gauche/droit est coupé et la touche doit donc être RELACHEE pendant la sonorisation ou l'enregistrement.

⑬ *TOUCHES DE SELECTION ECOUTE*

Ces touches permettent d'écouter au casque ou sur les enceintes, et de visualiser sur l'afficheur les signaux deux-pistes, Mono, Mix ou Groupes, individuellement ou combinés. **NOTE** : Si aucune touche n'est enfoncée, aucun signal ne sera écouté ou visualisé !

NON USARE microfoni sbilanciati con l'alimentazione phantom inserita poichè potrebbero rimanere danneggiati. I microfoni dinamici bilanciati possono generalmente esser usati con l'alimentazione phantom inserita (fare riferimento al manuale del microfono).

Prima di inserire i 48V bisogna collegare i microfoni e abbassare tutti i fader per non danneggiare gli apparecchi esterni.

⑧ *CUFFIE*

L'uscita PHONES è su prese jack da 1/4" di tipo "A" a 3-poli adatta a cuffie con impedenza da 200W o superiori.

⑨ *TB LEVEL*

E' previsto un ingresso bilanciato per il microfono di Talkback. Il segnale può essere indirizzato a Aux 1/2 o 3/4 (generalmente per il foldback dell'artista) o al Mix L/R premendo i relativi tasti. Il livello del segnale è regolato dal controllo TB LEVEL.

⑩ *RITORNI STEREO (STEREO RETURNS)*

Sono disponibili due Ritorni Stereo per le uscite di effetti. Essi sono mixati direttamente sui bus Mix L/R ad un livello impostato con RET-1 o RET-2. Se viene usata una sorgente mono il jack di sinistra manda il segnale automaticamente a destra e sinistra.

⑪ *2TK LEVEL*

Questa manopola regola il livello dell'ingresso 2 Track, che è indirizzato alle cuffie, alle uscite monitor e ai meter o direttamente alle uscite Mix premendo il tasto relativo (12). Questi ingressi sono sbilanciati, su connettori RCA phono, e sono indicati per collegare il playback di un registratore per il monitoraggio.

⑫ *2TK SOSTITUISCE L'USCITA MIX!*

Esattamente! Quando questo tasto è premuto l'uscita Mix viene mandata all'ingresso 2TK ed è il modo ideale di diffondere musica registrata attraverso le uscite principali in applicazioni live senza usare altri canali d'ingresso. Per esempio, con il tasto inserito ed un lettore CD collegato, il fonico può regolare i livelli, l'EQ, parlare ai musicisti e monitorare ogni parte del mix (compreso il Mix finale) senza agire sul suono udito dagli spettatori. Quando il tasto viene rilasciato le uscite Mix tornano al mixer e il lettore CD viene escluso.

ATTENZIONE: La pressione di questo tasto esclude il segnale Mix L/R e quindi non va usato durante l'esibizione o la registrazione.

⑬ *SELEZIONE SORGENTE MONITOR*

Questi tasti permettono la scelta di 2TK, Mono, Mix o Gruppi come sorgente per Cuffie, uscite monitor e meter e possono essere scelte singolarmente o combinate. **NOTA**: Se NESSUN tasto viene premuto, non ci sarà alcun segnale sui meters o sui monitor!

NO USE micrófonos no balanceados con la alimentación 48V activada, ya que pueden estropearse por la tensión phantom. Pueden usarse normalmente micrófonos dinámicos balanceados con la alimentación activada (contacte con el fabricante de su micrófono para saberlo)

Los micrófonos deben estar conectados y los faders al mínimo al activar la alimentación phantom para evitar daños a los equipos externos.

⑧ *PHONES*

La salida PHONES (auriculares) aparece en un jack de 1/4" de tres polos, adecuada para auriculares con una impedancia de 200Ω o mayor.

⑨ *TB LEVEL*

Se ofrece una entrada balanceada para un micrófono de talkback. La señal puede dirigirse selectivamente a los auxiliares 1/2 o 3/4 (que pueden usarse típicamente para el foldback de los músicos) o a la mezcla L/R pulsando los conmutadores apropiados. El nivel de señal se ajusta con el control TB LEVEL

⑩ *RETORNOS ESTEREO*

Dispone de dos retornos estéreo balanceados para las salidas de unidades de efectos, que se mezclan directamente a los buses MIX L/R a un nivel ajustado por los controles RET-1 y RET-2. Si se usa una fuente mono, al conectarla sólo al jack izquierdo se dirige automáticamente esta señal a la izquierda y la derecha.

⑪ *NIVEL 2TK*

Este control rotativo ajusta el nivel de la entrada de cinta 2-pistas, que se dirige a los auriculares, salida de monitores y medidores. Estas entradas no balanceadas, en conectores RCA, son ideales para conectar el equipo de cinta para monitorización.

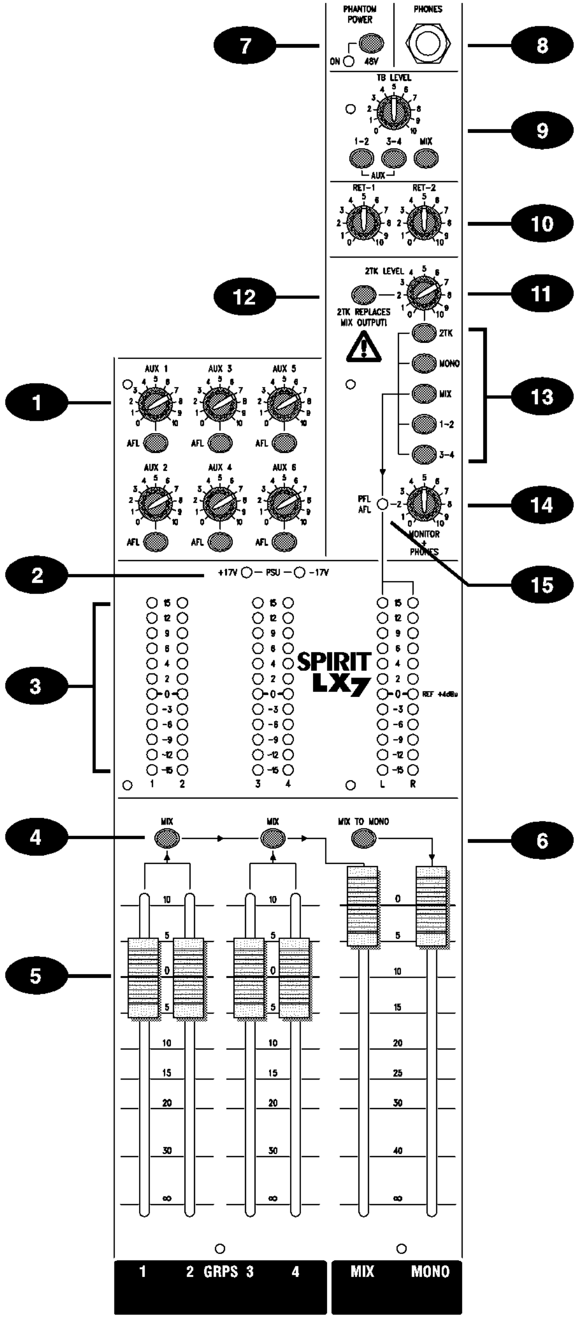
⑫ *2TK REEMPLAZA A LA SALIDA DE MEZCLA*

¡Este conmutador hace lo que dice! Al pulsarlo, la salida de mezcla se conmuta a la entrada de 2 pistas, y es una forma ideal de enviar música a las salidas principales antes de la actuación en aplicaciones de directo sin tener que usar unos valiosos canales de entrada. Por ejemplo, con el botón pulsado y un CD conectado y reproduciendo música antes de la actuación, el técnico puede estar ajustando niveles de canal, ecualizaciones, hablando con los músicos y monitorizando cualquier parte de la mezcla (incluyendo la mezcla principal) sin afectar el sonido de la audiencia. Al soltar este conmutador se vuelve a conectar la salida de mezcla al mezclador y se corta el CD.

PRECAUCION: Al pulsar este conmutador se corta la señal normal de mezcla L/R y por ello NO debe usarse durante una actuación en directo o en grabación.

⑬ *SELECTOR DE FUENTE DE MONITOR*

Estos conmutadores permiten la selección de 2TK, mono, mezcla o grupos como fuente de los auriculares, las salidas de monitor y los medidores, y pueden seleccionarse individualmente o en combinación. **NOTA:** Si no hay botones pulsados, no habrá señal en los medidores ni en los monitores.



⑭ MONITOR + PHONES

This control sets the output level to the MONITOR LEFT & RIGHT outputs. If Headphones are plugged into the PHONES jack the Monitor outputs are cut off, and the knob then sets the headphone listening level. When the PHONES are unplugged the Monitor output is restored. The source for the Monitor and Phones is selected by the five switches (13) above the control.

When any PFL or AFL switch is pressed the source for the Monitor outputs and Headphones is switched to the mono PFL/AFL signal without interrupting the other outputs from the mixer, to allow individual signals to be monitored. The original monitor source is restored when the PFL/AFL switches are released.

⑮ PFL/AFL

The PFL/AFL LED illuminates to show that a PFL/AFL is active and is the source for the monitors and meters. The LED will normally be OFF

ACHTUNG! Wenn dieser Schalter gedrückt wird, wird das normale Mix L/R-Signal unterbrochen. Aus diesem Grund sollte der Schalter nicht während einer Live-Veranstaltung oder einer Aufnahme gedrückt werden!

⑬ MONITOR SOURCE SELECT

Dieser Schalter ermöglicht das Anwählen der 2TK-, Mono, Mix- oder Gruppen-Signale als Quelle für den Phones- und den Monitor-Ausgang, sowie für die LED-Anzeige. Die Quellen können einzeln oder in Kombination geschaltet werden. Ist kein Schalter gedrückt, liegt auch kein Signal am Monitor-Ausgang und an der LED-Anzeige an.

⑭ MONITOR + PHONES

Mit diesem Regler wird der Ausgangspegel der Monitor Links- und Rechts-Ausgängen eingestellt. Ist ein Kopfhörer am PHONES-Anschluß eingesteckt, werden die Monitor-Ausgänge stummgeschaltet. Mit dem Regler wird dann der Pegel des Kopfhörers eingestellt. Wird der Kopfhörer wieder abgezogen, wird die Stummschaltung der Monitor-Ausgänge wieder aufgehoben. Die Quelle für die Monitor- und Phones-Ausgänge wird mit den 5 Schaltern (13) über dem Regler angewählt. Ist irgendein PFL- oder AFL-Schalter gedrückt, wird die Quelle für die Monitor-Ausgänge und der Kopfhörer-Ausgang zum Mono PFL/AFL-Signal geschaltet, ohne einen anderen Ausgang des Mixes zu unterbrechen. Einzelne Signale können jetzt abgehört werden. Die Orginal-Quelle ist wieder hörbar, wenn der PFL/AFL-Schalter erneut gedrückt wird.

⑮ PFL/AFL

Die PFL/AFL LED leuchtet, wenn ein PFL- oder AFL-Schalter auf dem LX7 gedrückt ist und die entsprechende Quelle auf die Monitor-Ausgänge und LED-Anzeige geschaltet ist. Normalerweise sollte die LED aus sein.

⑭ MONITOR + PHONES

Ce potentiomètre règle le niveau des sorties d'écoute gauche & droite. Si un casque est branché, les sorties Monitor seront coupées et le potentiomètre règle alors le niveau du casque. Lorsqu'aucun casque n'est branché, la sortie Monitor est restituée. La source d'écoute est sélectionnée à l'aide des cinq touches situées au-dessus du potentiomètre.

Lorsqu'une touche AFL ou PFL est enfoncée, le casque et l'afficheur reçoivent le signal mono PFL/AFL sans affecter les autres sorties de la console, cela permet de vérifier la présence et la qualité d'un signal. Lorsque les touches AFL/PFL sont relâchées, l'afficheur et le casque recoivent à nouveau la source initiale.

⑮ Led PFL/AFL

Cette Led s'illumine pour indiquer qu'une pré-écoute PFL/AFL est active. La Led doit normalement être éteinte.

⑭ MONITOR + CUFFIE

Questo controllo regola il livello di uscita alle uscite MONITOR DESTRO & SINISTRO. Se le cuffie sono collegate alla presa PHONES, le uscite Monitor sono escluse, e quindi la manopola regola il livello delle cuffie. Quando si toglie il jack della cuffie dalla presa PHONES l'uscita Monitor è ripristinata. La sorgente di Monitor e Phones viene selezionata dai cinque switch (13) sopra la regolazione. Quando uno dei tasti PFL o AFL è premuto la sorgente delle uscite Monitor e cuffie è commutata sul segnale mono PFL/AFL senza che le altre uscite dal mixer siano interrotte, per permettere ai segnali di essere monitorati individualmente. La sorgente originale viene ripristinata quando i tasti PFL/AFL sono rilasciati.

⑮ PFL/AFL

Il led PFL/AFL si illumina per indicare che un PFL/AFL è attivo e viene usato come sorgente per monitor e meters. Normalmente è spento.

⑭ NIVEL DE AURICULARES Y MONITORES

Este control ajusta el nivel de las salidas izquierda y derecha de monitores. Si hay unos auriculares conectados, se cortan las salidas de monitor, y el control ajusta el nivel de auriculares. Al desconectar los auriculares se restituye la salida de monitor.

Cuando se pulsa un botón PFL o AFL, la fuente para las salidas de monitor y auriculares se cambia por la señal mono PFL/AFL sin interrumpir las otras salidas del mezclador para permitir la monitorización de señales individuales. El LED PFL/AFL se ilumina para indicar que la fuente de los monitores y los medidores es una señal PFL/AFL. La fuente original de monitores se restituye al soltar los botones PFL/AFL.

⑮ PFL/AFL

El LED de PFL/AFL se ilumina para mostrar que hay un PFL/AFL activo y que es la fuente de los monitores y los medidores. El LED normalmente está apagado.

Setting Up & Troubleshooting

Initial Set Up

Once you have connected up your system (see the sections on connection and wiring earlier in this manual for guidance) you are ready to set initial positions for the controls on your mixer.

The front panel drawing on page 29 shows typical initial control positions which may serve as a useful guide to setting up the mixer for the first time.

Set up individual input channel as follows:

- Connect your sources (microphone, keyboard etc.) to the required inputs and release the MUTE switches. Note: Phantom powered mics should be connected before the 48V is switched on.
- Set Master faders at 0, input faders at 0, route the channels to MIX and set power amplifier level to suit the application.
- Provide a typical performance level signal and press the PFL button on the first channel, monitoring the level on the bargraph meters.
- Adjust the input gain until the meter display is in the amber section, with occasional peaks to the first red LED at a typical maximum source level. This allows sufficient headroom to accommodate peaks and establishes the maximum level for normal operation (but see note below).
- Repeat this procedure on other channels as required.
- Listen carefully for the characteristic sound of `feedback'. If you cannot achieve satisfactory input level setting without feedback, check microphone and speaker placement and repeat the exercise. If feedback persists, it may be necessary to use a Graphic Equaliser to reduce the system response at particular resonant frequencies.

Note: The initial settings should only be regarded as a starting point for your mix. It is important to remember that many factors affect the sound during a live performance, for instance the channel EQ settings or even the size of the audience!

You are now ready to start building the mix and this should be done progressively, listening carefully for each component in the mix and watching the meters for any hint of overload. If this occurs, back off the appropriate Channel Fader slightly until the level is out of the red segments, or adjust the Mix Master Faders. This procedure will ensure that the mixer is set up correctly, with adequate headroom. If more amplification is needed, adjust the power amplifier level controls.

Troubleshooting

No Power

- Is the mains supply present? Check that mains voltage selection is correct for your country.
- Is the mains lead firmly connected?
- Check the mains fusing
- If only one of the power indicators is illuminated, consult your Spirit dealer

Condenser Mic Not Working

- Is the 48V turned on?
- Is the mic plugged into the Mic input?
- Is the mic cable a balanced 3-wire type?

Meters not showing any signal

- Has the input gain been set correctly? (see above.)
- Is the source connected to the appropriate input socket for the level of signal?
- Do you have something conneceted on the Inserts, and is that external device switched on?
- Are the Master faders set at max., are input faders set high enough and is the channel routed to the output being monitored?
- Is the MUTE switch released on the relevant channels?
- Is the appropriate monitor select switch pressed?
- Is there a PFL/AFL pressed on another channel?

No Mix output

- Check that the Mix Master Fader is up?
- Check that the 2TK REPLACES MIX switch is released?

No Monitor output

- Is a headphone jack plugged in?
- Is the Monitor + Phones control set high enough?
- Is the appropriate monitor select switch pressed?

Headphones Distorting

- Are the headphones less than 200Ω impedance?
- Is the Monitor + Phones level set too high?

Richtiger Einsatz und Fehlersuche

Anfangseinstellungen

Wenn alles Equipment Ihres Systems angeschlossen ist (siehe hierzu im Abschnitt 'Vorbereitungen'), ist das System einsatzbereit, um erste Einstellungen an den Reglern vorzunehmen.

Die Abbildung auf Seite 29 zeigt die typischen ersten Einstellungen der Regler an. Diese Abbildung ist eine nützliche Anweisung für die erstmalige Mixer-Einstellung.

Folgendermaßen werden die einzelnen Eingangskanäle eingerichtet:

- Quelle an die entsprechenden Eingänge anschließen (z.B. Mikrofon, Keyboards, etc.). Mute-Funktionen alle aufheben. Mit Phantom-Spannung betriebene Mikrofone anschließen, bevor die 48V eingeschaltet wird.
- Master-Fader und Kanal-Eingangs-Fader auf 0 stellen; Kanäle zum MIx routen und die Power-Amps-Pegel entsprechend der Anwendung einpegeln.
- Einen typischen Auftritts-Signal-Pegel einstellen und den PFL-Schalter des ersten Kanals drücken. Den Signalpegel auf der LED-Anzeige beachten.
- Die Vorverstärkung der einzelnen Eingangskanäle so einstellen, bis das die LED-Anzeige das Signal im amberfarbenden Bereich anzeigt. Hierbei darf die erste rote Peak-LED leuchten. Dies ist ein typischer maximaler Signalpegel. Dadurch stellt sich ein genügend großer Headroom (Verstärungsreserve) für eine normale Beschallung ein (Beachten Sie jedoch auch den nachfolgenden Hinweis).
- Diese Vorgehensweise auch auf den anderen Kanälen anwenden.
- Achten Sie auf den charakteristischen Sound einer beginnenden Rückkopplung. Kann keine befriedigende Einstellung erfolgen, ohne daß Rückkopplungen entstehen, müssen Mikrofon- und Lautsprecher-Aufstellung nochmals geprüft werden. Bei weiterhin anhaltenden Rückkopplungen kann der Einsatz eines Graphic Equalizers notwendig werden, um den System-Frequenzgang bei bestimmten Resonanz-Frequenzen abzuschwächen.

Réglages et problèmes de fonctionnement

Réglages initiaux

Une fois votre système connecté (voir plus haut les sections concernant la connexion et le câblage), vous êtes prêts pour les réglages initiaux.

Le dessin de la face page 29 vous montre la position typique des potentiomètres, cela peut être une aide précieuse pour régler votre console la première fois. Réglez les voies d'entrée comme suit :

- Connectez l'entrée requise (microphones, claviers...). Note : les micros alimentés en fantôme doivent être connectés avant que le 48 V soit mis en route.
- Réglez les généraux et les faders d'entrée à 0, réglez les niveaux des amplificateurs de puissance à environ 70%.
- Injectez un signal de niveau typique et appuyez sur la touche PFL de la première voie d'entrée. Le niveau du signal sera visualisé sur l'afficheur.
- Ajustez le gain d'entrée jusqu'à ce que le VU-mètre atteigne la partie orange (0dB) et ceci pour un niveau de source maximum. Cela permet une dynamique suffisante pour que les crêtes soient supportées et que l'on travaille à un niveau de fonctionnement normal (voir note ci-dessous).
- Répétez cette procédure sur les autres voies utilisées. Avec l'addition de nouvelles voies d'entrée, les VU-mètres peuvent très bien atteindre la zône rouge. Dans ce cas, ajustez le niveau général en utilisant les faders du Master.
- Ecoutez avec attention les caractéristiques du son afin d'éviter le Larsen. Si vous ne pouvez pas atteindre des niveaux corrects et satisfaisants sans Larsen, vérifiez la position des micros et des haut-parleurs et refaites vos réglages. Si le Larsen persiste, il peut être nécessaire d'utiliser un égaliseur graphique pour corriger la réponse en fréquence du système.

Set Up & Guida all'individuazione dei problemi

Set Up Iniziale

Una volta che il sistema è collegato (vd. capitolo relativo) siete pronti ad impostare la posizione iniziale dei controlli.

Il disegno del pannello frontale a pagina 18 mostra un posizione tipica dei controlli che può essere usata come guida per la prima volta che si effettua tale operazione.

Impostate i singoli canali d'ingresso come segue:

- Collegate le sorgenti (microfono, tastiere, etc.) agli ingressi desiderati e rilasciate i tasti MUTE. Nota: I microfoni con alimentazione phantom vanno collegati prima che +48V sia inserito.
- Impostate i Master Fader e quelli di ingresso a 0, indirizzate i canali al MIX e portate l'amplificatore di potenza ad un livello adeguato all'installazione.
- Date un livello tipico del segnale dell'esecuzione e premete PFL sul primo canale, controllando il livello sui meters.
- Regolate il gain d'ingresso finchè il meter raggiunge la zona arancione, con sporadici picchi sul primo led rosso come massimo. Questo garantisce un headroom sufficiente per sistemare i picchi e stabilisce il livello massimo per l'uso normale.
- Ripetete la procedura con gli altri canali secondo necessità.
- Ascoltate attentamente il tipico suono del "feedback". Se non riuscite ad avere un livello d'ingresso soddisfacente senza feedback, controllate la posizione dei microfoni e dei diffusori, quindi ripetete la prova. Se il feedback continua, potrebbe essere necessario usare un equalizzatore grafico per ridurre la risposta del sistema a particolari frequenze.

Ajustes y Solución de Problemas

Ajuste Inicial

Una vez haya conectado su sistema (vea las secciones de conexionado y cableado en este manual) ya está preparado para ajustar los controles de su mezclador a sus posiciones iniciales.

El dibujo del panel frontal en la página 29 muestra las posiciones iniciales típicas de los controles, lo que puede resultar una guía útil para ajustar el mezclador la primera vez.

Ajuste los canales de entrada como sigue:

- Conecte sus fuentes (micrófono, teclado, etc.) a las entradas y desactive los botones MUTE. Nota: Los micrófonos alimentados con phantom deben conectarse antes de activar los 48V.
- Sitúe los faders master a 0, los faders de entrada a 0 direccione los canales a MIX y ajuste los niveles de la etapa de potencia sobre el 70%.
- Inyecte una señal de nivel típico de actuación y pulse el botón PFL en el primer canal, monitorizando el nivel en los medidores bargraph.
- Ajuste la ganancia de entrada hasta que el medidor esté en la sección ámbar, con picos ocasionales al primer LED rojo al nivel típico máximo de la fuente. Esto permite suficiente techo dinámico para soportar picos y establece el nivel máximo para trabajo normal (pero vea la nota más abajo).
- Repita este procedimiento en los canales que necesite.
- Escuche atentamente el sonido característico de la realimentación. Si no puede alcanzar el nivel de entrada deseado sin realimentaciones, compruebe las posiciones de los micrófonos y los altavoces y repita el ejercicio. Si la realimentación persiste, puede ser necesario usar un ecualizador gráfico para reducir la respuesta del sistema a frecuencias resonantes particulares.

HINWEIS: Diese Ersteinstellungen sollten als Anfangs-Einstellungen Ihres Mix angesehen werden. Es ist wichtig, zu erkennen, daß viele Faktoren den Sound während einer Live-Darbietung beeinflussen, z.B. die EQ-Einstellungen der einzelnen Kanäle oder auch die Größe des Publikums.

Jetzt können Sie mit dem Aufbau des Mixes beginnen und dies sollte schrittweise so erfolgen, indem Sie aufmerksam jeden Anteil des Mix genauestens abhören und auch die Aussteuerung an der LED-Anzeige auf Übersteuerung hin überwachen. Sollte eine Übersteuerung eintreten, den entsprechenden Kanal-Fader langsam zurückpegeln, bis der Pegel aus dem roten Anzeigebereich läuft. Oder den Master-Fader zurücknehmen.

Wenn Sie diese Vorgehensweise einhalten, können Sie sicher sein, daß der Mixer richtig konfiguriert ist und genügend Headroom vorhanden ist. Sollte noch etwas mehr Verstärkung benötigt werden, regeln Sie die angeschlossenen Power-Amps nach.

Fehlersuche

Keine Netzspannung

- Prüfen, ob Netzgerät richtig angeschlossen ist.
- Prüfen Sie auch, ob die örtliche Netzspannung mit der eingestellten übereinstimmt.
- Überprüfen Sie die Hauptsicherungen
- Falls nur eine Spannungsanzeige auf dem LX7 leuchtet, dann verständigen Sie Ihren Händler

Kondensator-Mikrofon funktioniert nicht

- Ist der 48V-Schalter betätigt, Stellung ON ?
- Ist das Mikrofon ordnungsgemäß am Mikrofon-Anschluß eingesteckt?
- Ist das Mikrofon-Kabel ein symmetrisches 3-adriges Kabel?

LED-Anzeigen zeigen kein Signal an

- Ist die Vorverstärkung richtig eingestellt?
- Ist die Quelle auch an dem Eingang eingesteckt, der für diesen Signalpegel vorgesehen ist?
- Ist am Insert (Einschleifpunkt) ein Stecker eingesteckt und ist das eingeschleifte Gerät eingeschaltet?
- Sind die Master-Fader auf Maximum geregelt?
- Sind die Kanal-Fader hoch genug eingestellt und wird der Kanal zu den Monitor-Ausgängen geroutet?
- Sind die MUTE-Schalter der relevanten Kanäle nicht aktiviert?
- Ist der entsprechende Monitor-Schalter gedrückt?
- Ist ein PFL/AFL-Schalter auf einem anderen Kanal gedrückt?

Kein Mix-Ausgangssignal

- Sind die Mix-Master-Fader richtig gepegelt?
- Ist der 2TK REPLACES MIX OUTPUT-Schalter nicht gedrückt?

Kein Monitor-Ausgang

- Ist ein Kopfhörer am Mischpult eingesteckt?
- Ist der Monitor- und Phones-Regler hoch genug eingestellt?
- Ist der entsprechende Monitor-Schalter gedrückt?

Kopfhörer-Verzerrungen

- Ist die Eingangs-Impedanz geringer als 200 Ω?
- Ist der Monitor- und Phones-Pegel zu hoch eingestellt?

Note : Les réglages initiaux doivent être considérés comme un point de départ pour votre mix. Il est important de rappeler que beaucoup de facteurs affectent le son durant un concert, par exemple la taille de l'audience .

Vous êtes prêts maintenant à démarrer votre mixage. Celui-ci doit être fait progressivement, en écoutant soigneusement chaque élément du mix et en surveillant les afficheurs. Si une surcharge apparait, baissez doucement le fader de la voie appropriée jusqu'à descendre sous la partie rouge, ou baissez le niveau général.

Problèmes de fonctionnement

Pas de tension présente

- Le courant secteur est-il bien présent - Vérifiez la prise avec un autre appareil.
- Le cordon secteur est-il bien connecté ?
- Vérifiez le fusible
- Si une seule Led de mise sous tension est allumée, consultez votre revendeur.

Le microphone ne fonctionne pas

- Le 48V est-il mis en service ?
- La prise est-elle bien branchée sur l'entrée micro ?
- Le câble micro est-il un câble symétrique trois conducteurs ?

Les afficheurs ne visualisent aucun signal

- Le gain d'entrée a-t-il été réglé correctement (voir au-dessus) ?
- La source est-elle bien connectée à l'embase correspondant au bon niveau de signal ?
- Avez-vous un appareil connecté en insert, cet appareil est-il bien en fonctionnement ?
- Les faders de mix sont-ils au maximum, et les faders de voie à une position suffisante ? La voie est-elle bien dirigée vers l'écoute ?
- La touche MUTE est-elle bien relâchée ?
- La touche de sélection d'écoute est-elle bien enfoncée ?
- Une touche PFL/AFL est-elle enfoncée sur une autre voie ?

Aucun signal en sortie mix

- vérifiez que les généraux sont en haut
- vérifiez que la touche 2TK REPLACES MIX est relâchée

Aucun signal en écoute

- La prise casque est-elle branchée ?
- Les potentiomètres écoute + casque sont-ils bien réglés ?
- La touche correspondante de sélection d'écoute est-telle bien enfoncée ?

Distorsion au casque

- le casque a-t-il une impédance supérieure à 200Ω?
- Le niveau écoute + casque est-il suffisant ?

Nota: Le impostazioni iniziali vanno considerate solo come punto di partenza. Va ricordato che molti fattori influenzano il suono durante una esecuzione dal vivo, ad esempio le quantità di pubblico.

Ora siete pronti a fare il mixaggio e questo va fatto per gradi, ascoltando attentamente ogni componente del mixaggio e osservando i meters per il sovraccarico. Se questo si verifica, abbassate adeguatamente il fader relativo finchè livello è fuori dal segmento rosso oppure regolate il Master Fader. Questo assicura un corretto set up. Se è necessaria maggiore amplificazione, regolare il livello dell'amplificatore.Quando vengono aggiunti canali al mixer i meters potrebbero arrivare al rosso. Regolate il livello generale con il Master Fader, se necessario.

NOTA: Il livello di ogni sorgente nell'uscita finale è influenzato da molti fattori, tra cui la regolazione di sensibilità (Sensitivity), il Channel Fader e gli Output Faders. Dovreste tentare di usare solo il gain necessario ad ottenere un buon bilanciamento tra i segnali, con i faders impostati come descritto sopra. Se il gain d'ingresso è troppo alto il fader andrà abbassato troppo per compensare ed avere abbastanza gioco per il mixaggio e c'è il rischio di feedback perchè piccoli spostamenti del fader hanno un effetto significativo sul livello di uscita. Se il gain è troppo basso non ce ne sarà abbastanza sui faders per alzare il segnale ad un livello adeguato.

Guida all'individuazione dei problemi

Non c'è alimentazione

- Controllare che ci sia tensione e che sia selezionata correttamente.
- Il cavo di alimentazione è collegato saldamente?
- Controllare i fusibili.
- Se si illumina solo uno degli indicatori di alimentazione rivolgetevi al rivenditore Spirit.

Il microfono a condensatore non funziona

- L'alimentazione phantom è inserita?
- Il microfono è collegato all'ingresso Mic?
- Il cavo microfonico è del tipo XLR bilanciato a 3 poli?

I meter non indicano alcun segnale

- Il gain è impostato correttamente?
- La sorgente è collegata all'ingresso adeguato al livello del segnale?
- C'è qualche apparecchio acceso e collegato agli Inserts?
- Controllare che i Master faders siano al massimo, i fader d'ingresso siano al livello adatto e che il canale sia indirizzato all'uscita che viene monitorata?
- Il tasto MUTE è disattivato sul canale in questione?
- E' stato selezionato il giusto monitor?
- C'è un PFL/AFL attivo su un altro canale?

Nessuna uscita Mix

- Controllare che il Master Fader Mix sia attivo?
- Controllare che il tasto 2TK REPLACES MIX non sia attivato?

Nessuna uscita Monitor

- C'è una cuffia inserita?
- Il livello Monitor + Phones è abbastanza alto?
- E' stato selezionato il giusto monitor?

Distorsione in cuffia

- Le cuffie hanno un'impedenza inferiore a 200Ω?
- Il livello Monitor + Phones è troppo alto?

Nota: Los ajustes iniciales sólo sirven como punto de comienzo en su mezcla. Es importante recordar que muchos factores pueden afectar al sonido durante una actuación en directo, como por ejemplo la ecualización de un canal o incluso la cantidad de personas en el local.

Ahora Ud. ya está preparado para empezar a construir la mezcla, escuchando cuidadosamente para cada componente en la mezcla y vigilando algún indicio de sobrecarga en los medidores. Si esto ocurre, baje ligeramente el fader del canal correspondiente hasta que el nivel deje de encender los segmentos rojos del medidor, o reajuste los faders master de mezcla. Este procedimiento le asegurará que el mezclador está correctamente ajustado, y con un techo dinámico adecuado. Si se requiere más amplificación, ajuste los controles del amplificador de potencia.

Solución de Problemas

No hay alimentación

- Hay corriente? Compruebe la toma de corriente y que el selector de voltaje esté en la posición correcta.
- Está el cable de alimentación firmemente conectado?
- Compruebe el estado del fusible
- Si sólo uno de los indicadores de alimentación está iluminado, consulte con su distribuidor Spirit

El micro de condensador no funciona

- Está la alimentación 48V de micrófono activada?
- Está el micro conectado en la entrada de micro?
- El cable del micro es del tipo balanceado de 3 cables?

Los medidores no muestran ninguna señal

- Tiene el Nivel del Ganancia ajustado correctamente? (ver arriba)
- Está la fuente conectada al conector de entrada apropiado por el nivel de señal?
- ¿Tiene algo conectado en las inserciones, y está ese dispositivo externo en marcha?
- Están los faders de Mix ajustado al máximo, y están los controles de volumen de entrada lo suficientemente altos?
- ¿Están desactivados los botones de MUTE en todos los canales utilizados?
- ¿Está pulsado el botón de selección de monitor apropiado ?
- Hay algún PFL/AFL pulsado en otro canal?

No hay salida de mezcla

- Compruebe que el fader master de mezcla está arriba.
- Compruebe que el botón 2TK REEMPLAZA A LA MEZCLA esté desactivado.

No hay salida en el amplificador de monitores

- Está el jack de auriculares insertado?
- ¿Está el control de nivel de monitor y auriculares suficientemente alto?
- ¿Está pulsado el conmutador de selección de monitor apropiado?

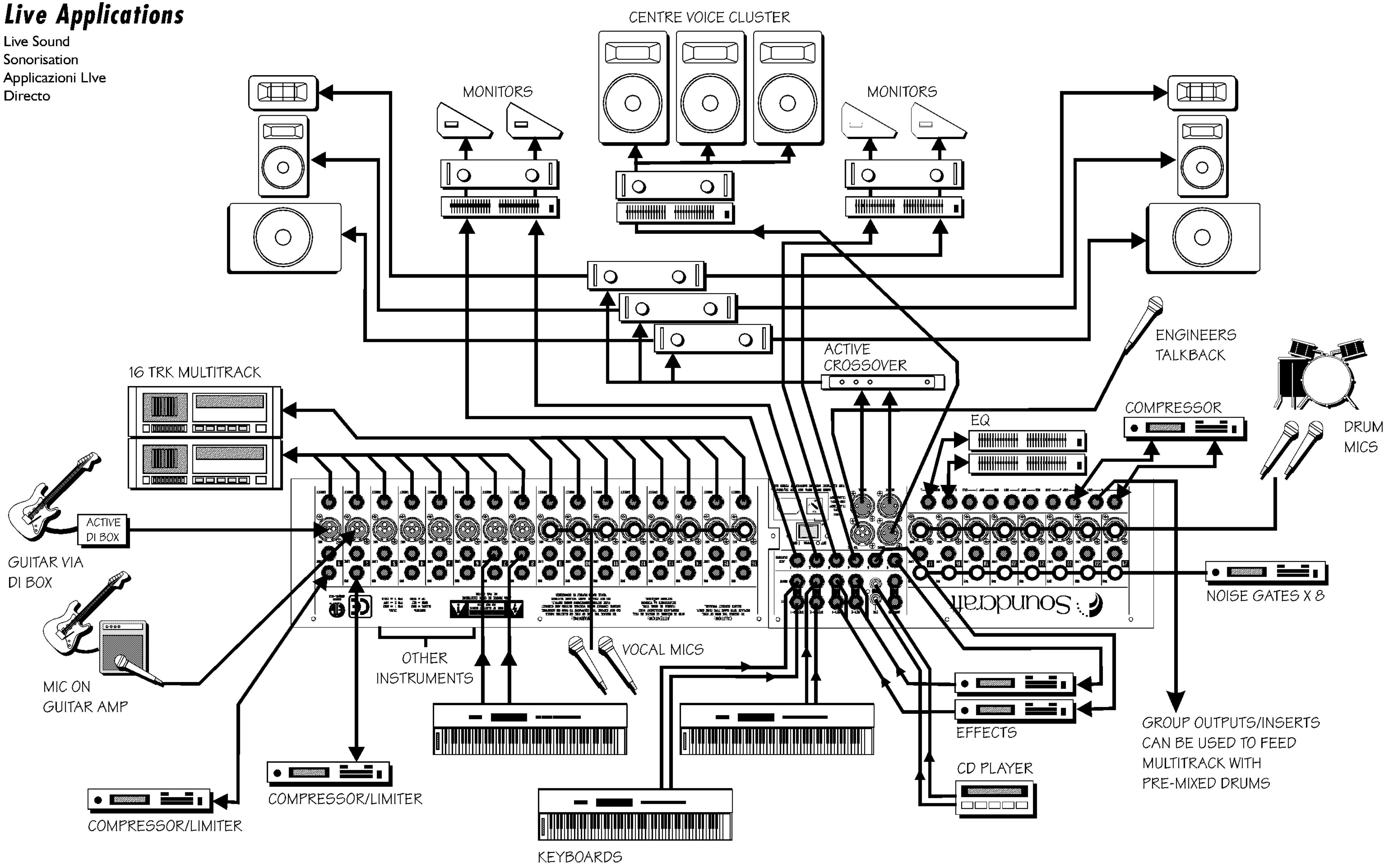
Distorsión de auriculares

- ¿Son los auriculares de impedancia mayor de 200Ω?
- ¿Está el control de nivel de monitor y auriculares suficientemente alto?

APPLICATIONS Anwendungen - Applications - Applicazioni - Aplicaciones

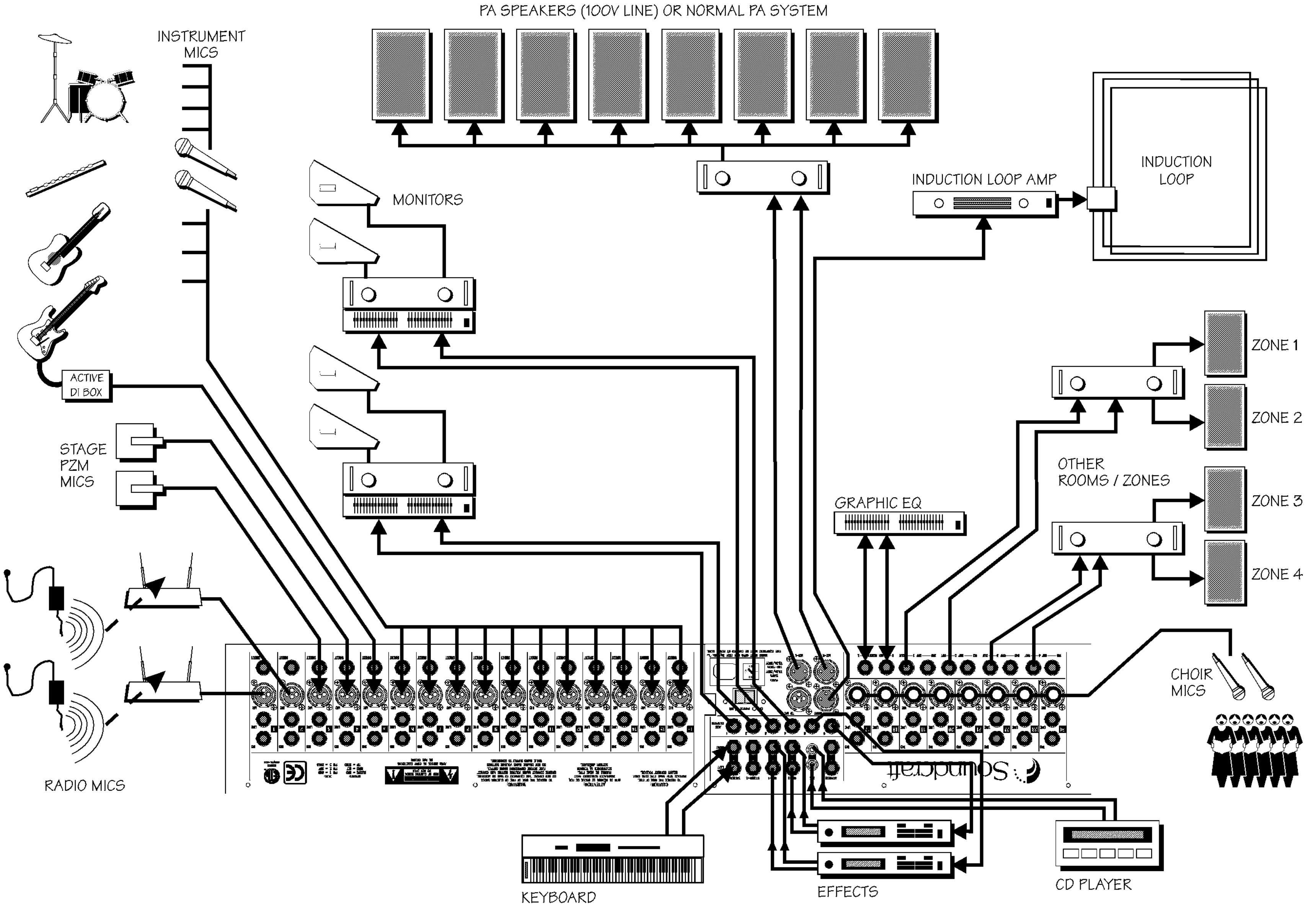
Live Applications

Live Sound
Sonorisation
Applicazioni LIve
Directo


CENTRE VOICE CLUSTER
MONITORS
MONITORS
16 TRK MULTITRACK
ACTIVE CROSSOVER
ENGINEERS TALKBACK
EQ
COMPRESSOR
DRUM MICS
ACTIVE DI BOX
GUITAR VIA DI BOX
MIC ON GUITAR AMP
OTHER INSTRUMENTS
VOCAL MICS
NOISE GATES X 8
EFFECTS
CD PLAYER
GROUP OUTPUTS/INSERTS CAN BE USED TO FEED MULTITRACK WITH PRE-MIXED DRUMS
COMPRESSOR/LIMITER
COMPRESSOR/LIMITER
KEYBOARDS
Soundcraft

Places of Worship

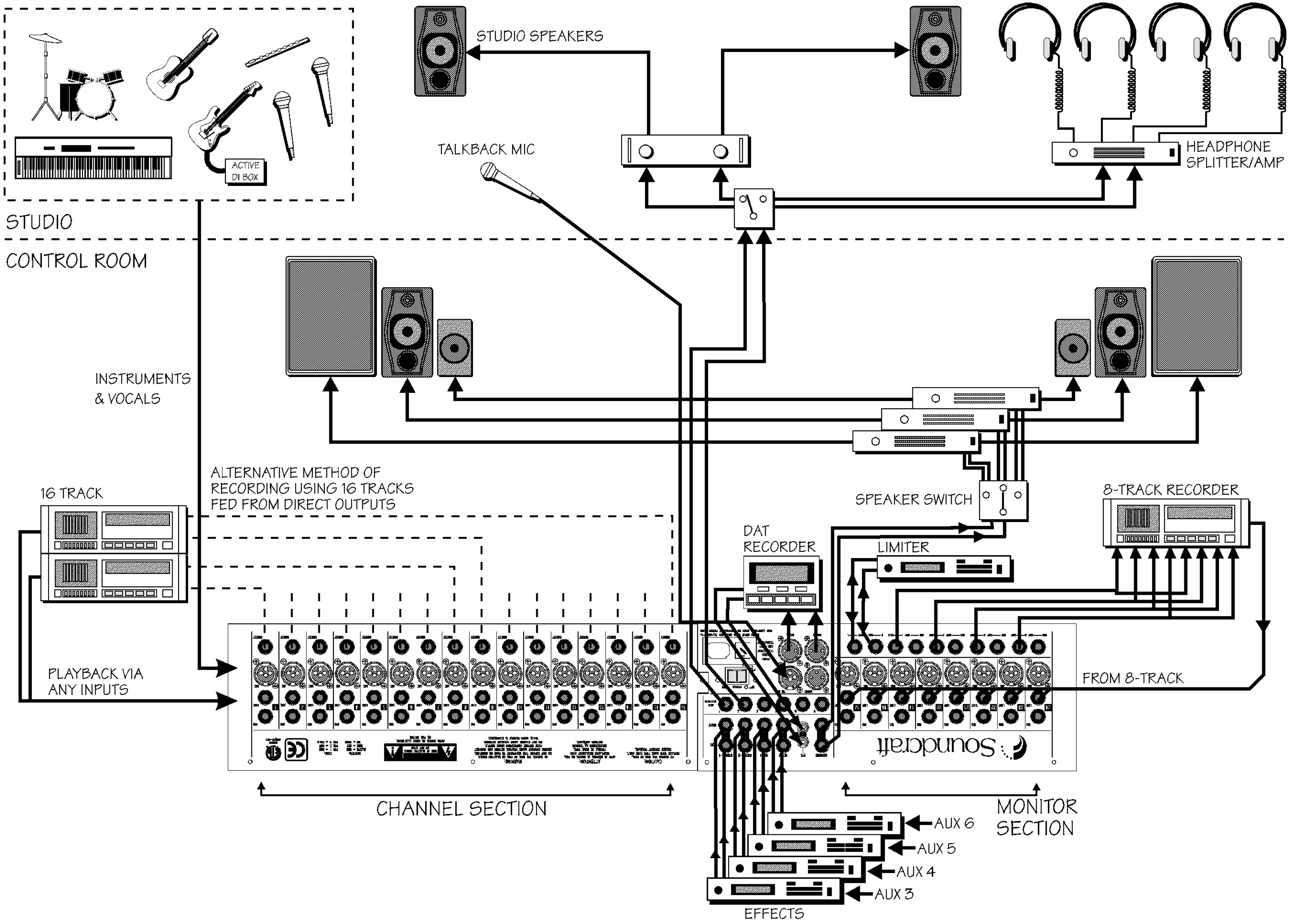
Kirchen
Lieux de culte
Luoghi di culto
Iglesias



PA SPEAKERS (100V LINE) OR NORMAL PA SYSTEM
INSTRUMENT MICS
MONITORS
INDUCTION LOOP AMP
INDUCTION LOOP
ACTIVE DI BOX
STAGE PZM MICS
ZONE 1
ZONE 2
OTHER ROOMS / ZONES
ZONE 3
ZONE 4
GRAPHIC EQ
CHOIR MICS
RADIO MICS
Soundcraft
KEYBOARD
EFFECTS
CD PLAYER

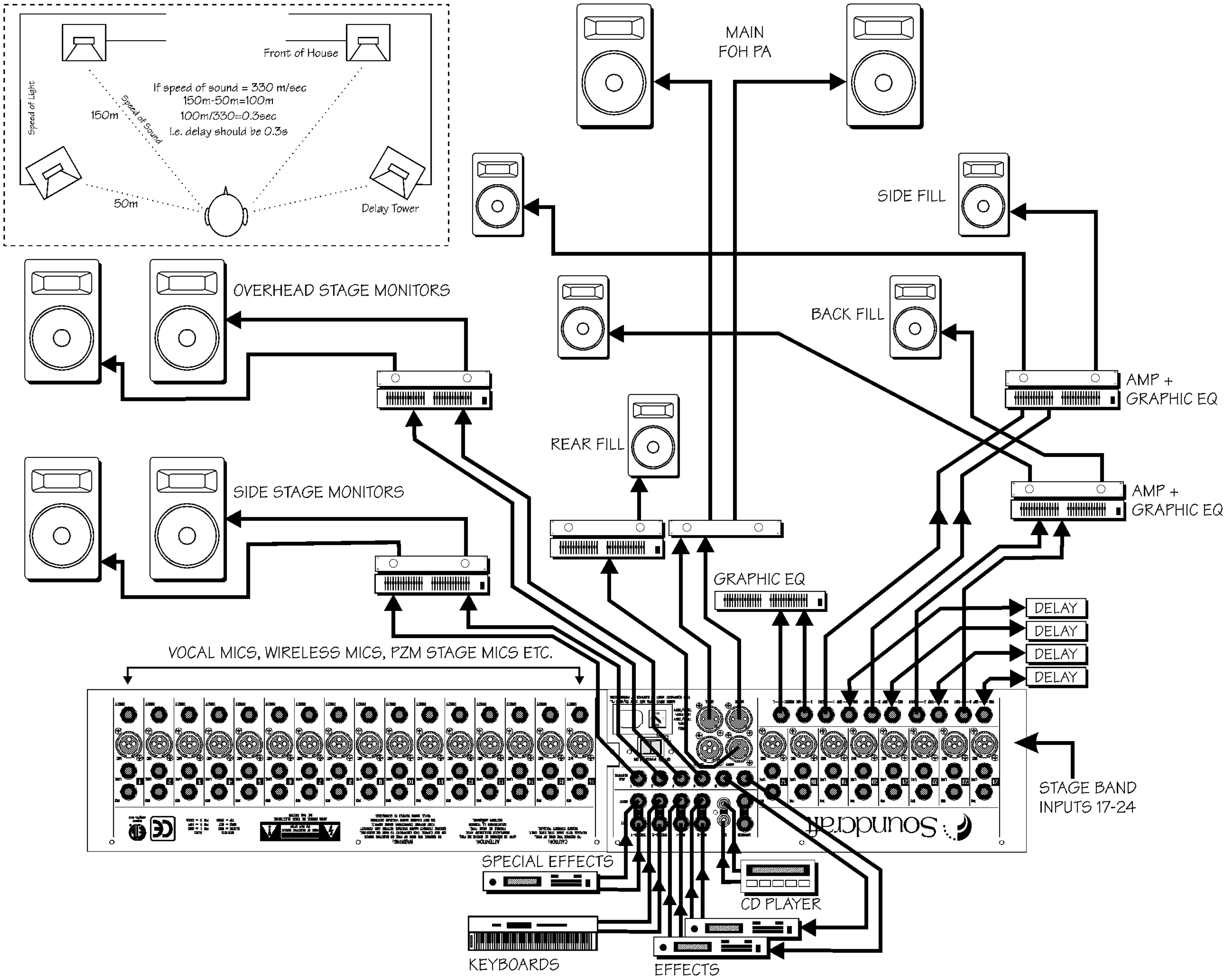
Recording Studio

Aufnahme-Studio
Stdio d'enregistrement
Studi di regristrazione
Estudio de grabación


ACTIVE DI BOX
STUDIO
CONTROL ROOM
STUDIO SPEAKERS
TALKBACK MIC
HEADPHONE SPLITTER/AMP
INSTRUMENTS & VOCALS
16 TRACK
ALTERNATIVE METHOD OF RECORDING USING 16 TRACKS FED FROM DIRECT OUTPUTS
PLAYBACK VIA ANY INPUTS
SPEAKER SWITCH
8-TRACK RECORDER
DAT RECORDER
LIMITER
FROM 8-TRACK
CHANNEL SECTION
MONITOR SECTION
AUX 6
AUX 5
AUX 4
AUX 3
EFFECTS

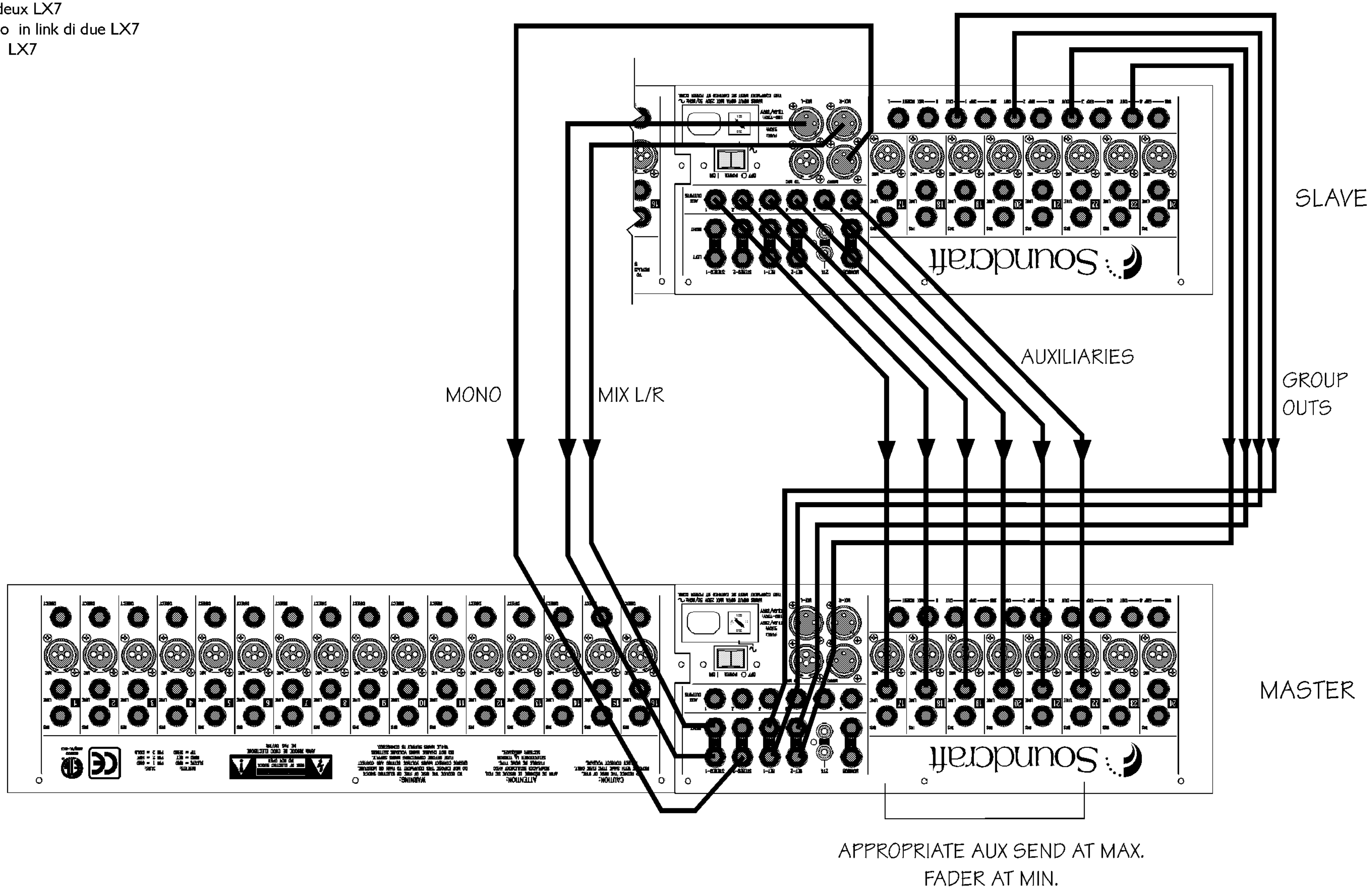
Theatre

Einsatz im Theater
Theatre
Teatro
Teatros


Front of House
Speed of Light
Speed of Sound
150m
50m
Delay Tower
If speed of sound = 330 m/sec
150m-50m=100m
100m/330=0.3sec
i.e. delay should be 0.3s
MAIN
FOH PA
SIDE FILL
BACK FILL
REAR FILL
OVERHEAD STAGE MONITORS
SIDE STAGE MONITORS
AMP +
GRAPHIC EQ
AMP +
GRAPHIC EQ
GRAPHIC EQ
DELAY
DELAY
DELAY
DELAY
VOCAL MICS, WIRELESS MICS, PZM STAGE MICS ETC.
STAGE BAND
INPUTS 17-24
Soundcraft
SPECIAL EFFECTS
CD PLAYER
KEYBOARDS
EFFECTS

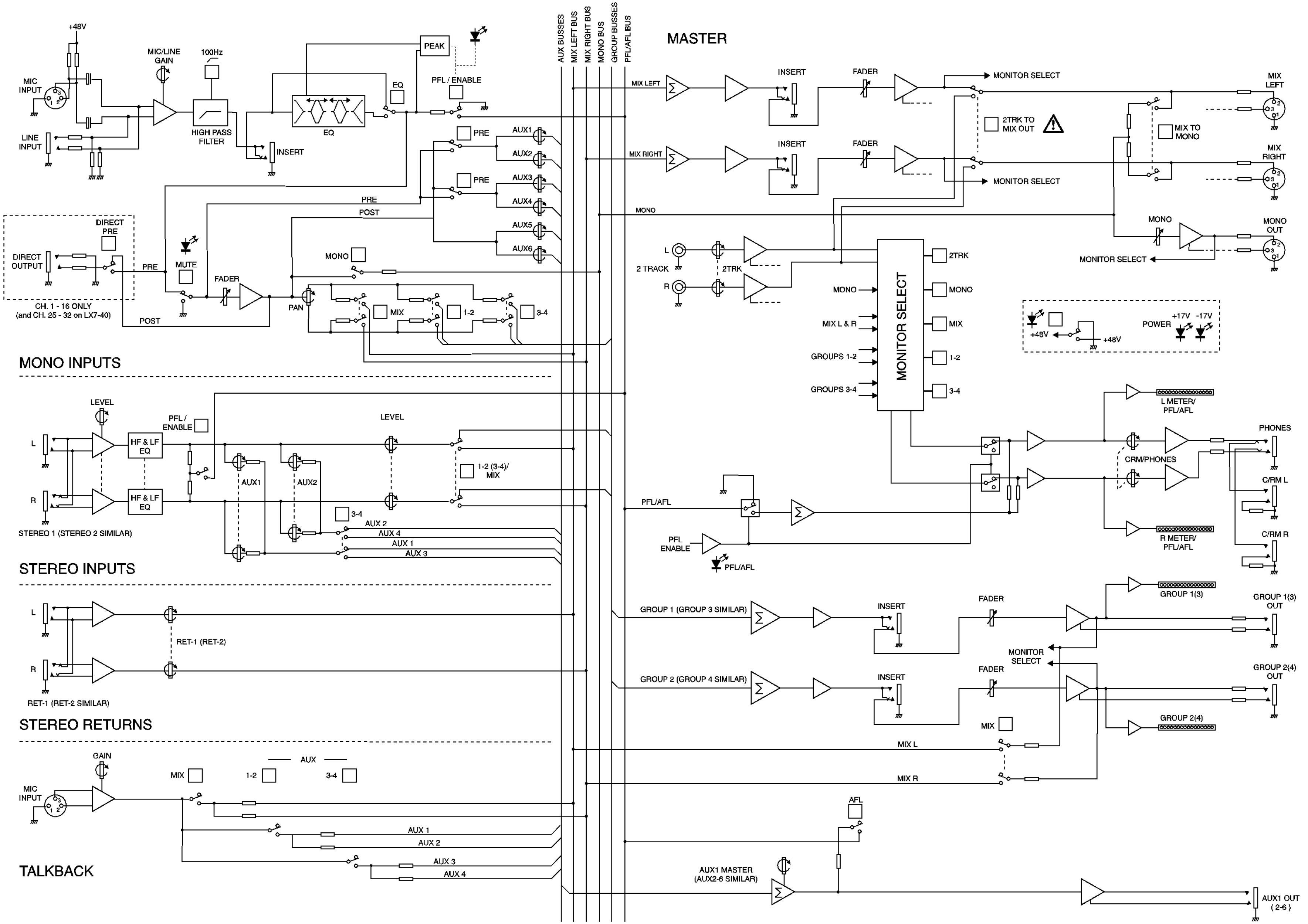
Linking two LX7s

Verbinden zweier LX7
Couplage de deux LX7
Funzionamento in link di due LX7
Enlace de dos LX7


SLAVE
AUXILIARIES
MONO
MIX L/R
GROUP
OUTS
MASTER
APPROPRIATE AUX SEND AT MAX.
FADER AT MIN.

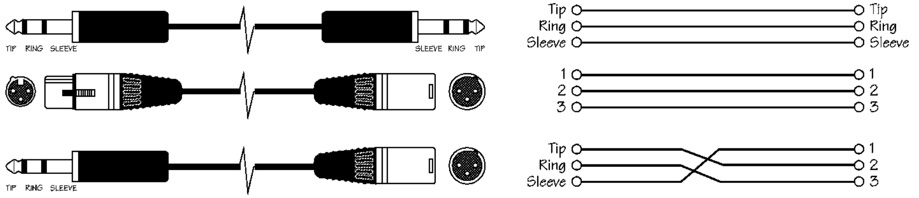
System Block Diagram

Block Diagramm / Synoptique / Diagramma a blocchi / Diagrama de Bloques del Sistema

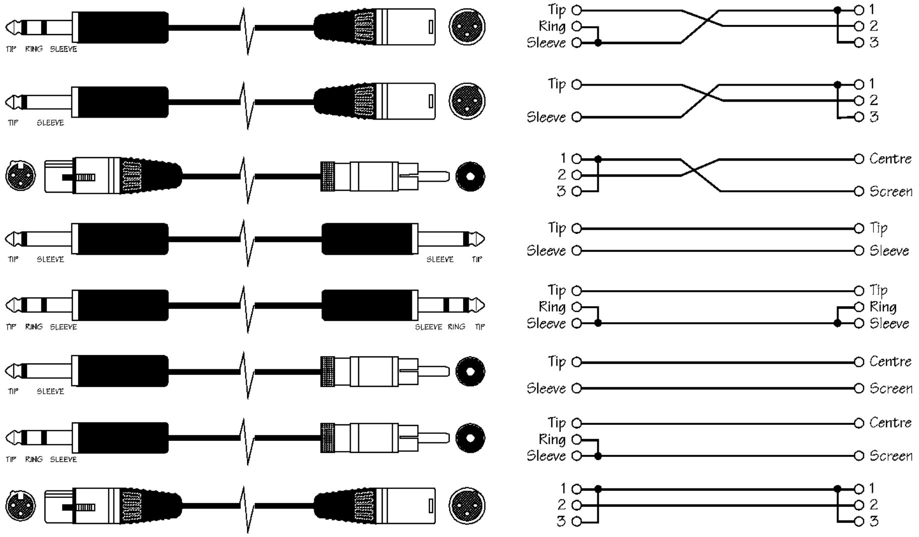

+48V
MIC INPUT
LINE INPUT
MIC/LINE GAIN
100Hz
HIGH PASS FILTER
INSERT
EQ
PEAK
PFL / ENABLE
PRE
AUX1
AUX2
AUX3
AUX4
AUX5
AUX6
POST
DIRECT PRE
DIRECT OUTPUT
CH. 1 - 16 ONLY
(and CH. 25 - 32 on LX7-40)
MUTE
FADER
MONO
PAN
MIX
1-2
3-4
MONO INPUTS
AUX BUSSES
MIX LEFT BUS
MIX RIGHT BUS
MONO BUS
GROUP BUSSES
PFL/AFL BUS
MASTER
MIX LEFT
MIX RIGHT
MONITOR SELECT
2TRK TO MIX OUT
MIX TO MONO
MIX LEFT
MIX RIGHT
MONO OUT
2 TRACK
L
R
2TRK
MIX L & R
GROUPS 1-2
GROUPS 3-4
2TRK
+48V
POWER
+17V
-17V
LEVEL
PFL / ENABLE
HF & LF EQ
1-2 (3-4)/ MIX
AUX 2
AUX 4
AUX 1
AUX 3
STEREO 1 (STEREO 2 SIMILAR)
STEREO INPUTS
L METER/ PFL/AFL
PHONES
CRM/PHONES
C/RM L
C/RM R
R METER/ PFL/AFL
PFL/AFL
PFL ENABLE
RET-1 (RET-2)
RET-1 (RET-2 SIMILAR)
STEREO RETURNS
GROUP 1 (GROUP 3 SIMILAR)
GROUP 2 (GROUP 4 SIMILAR)
GROUP 1(3)
GROUP 1(3) OUT
GROUP 2(4)
GROUP 2(4) OUT
MIX L
MIX R
TALKBACK
GAIN
AUX
AFL
AUX1 MASTER (AUX2-6 SIMILAR)
AUX1 OUT (2-6)

Connecting Leads / Ideale kabel und kabel-verbindungen / Connexions / Collegamenti / Conectando los cables

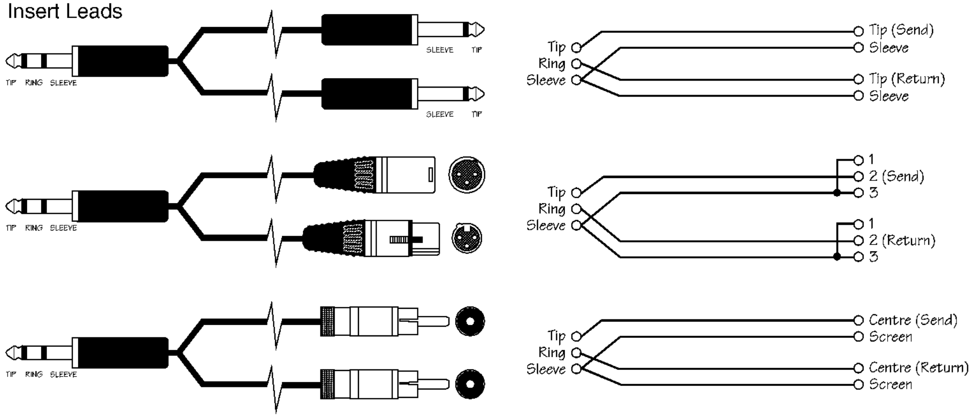
Balanced


TIP RING SLEEVE
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
1
2
3
1
2
3
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2
3

Unbalanced

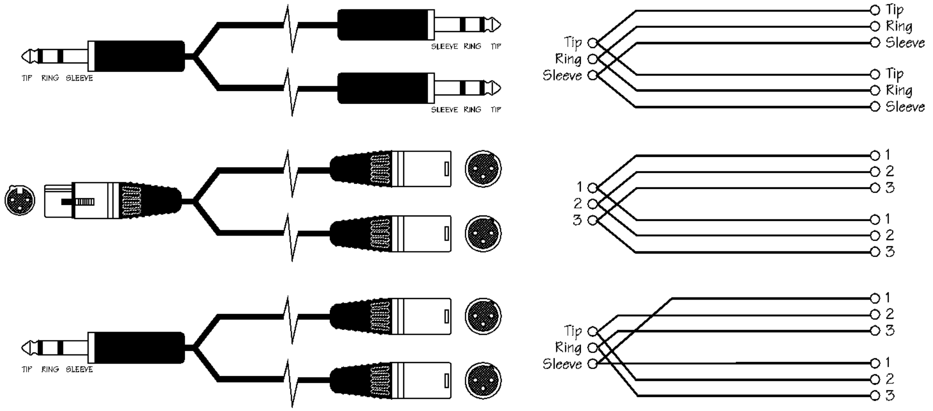

TIP RING SLEEVE
Tip
Ring
Sleeve
1
2
3
TIP SLEEVE
Tip
Sleeve
1
2
3
1
2
3
Centre
Screen
TIP SLEEVE
SLEEVE TIP
Tip
Sleeve
Tip
Sleeve
TIP RING SLEEVE
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
TIP SLEEVE
Tip
Sleeve
Centre
Screen
TIP RING SLEEVE
Tip
Ring
Sleeve
Centre
Screen
1
2
3
1
2
3

Insert Leads

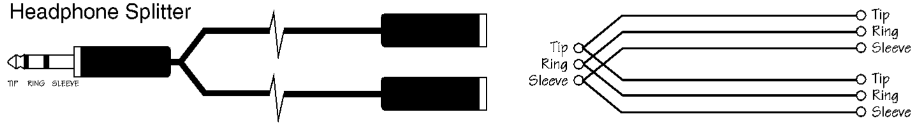

SLEEVE TIP
TIP RING SLEEVE
SLEEVE TIP
Tip
Ring
Sleeve
Tip (Send)
Sleeve
Tip (Return)
Sleeve
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2 (Send)
3
1
2 (Return)
3
TIP RING SLEEVE
Tip
Ring
Sleeve
Centre (Send)
Screen
Centre (Return)
Screen

'Y' Leads (Balanced)

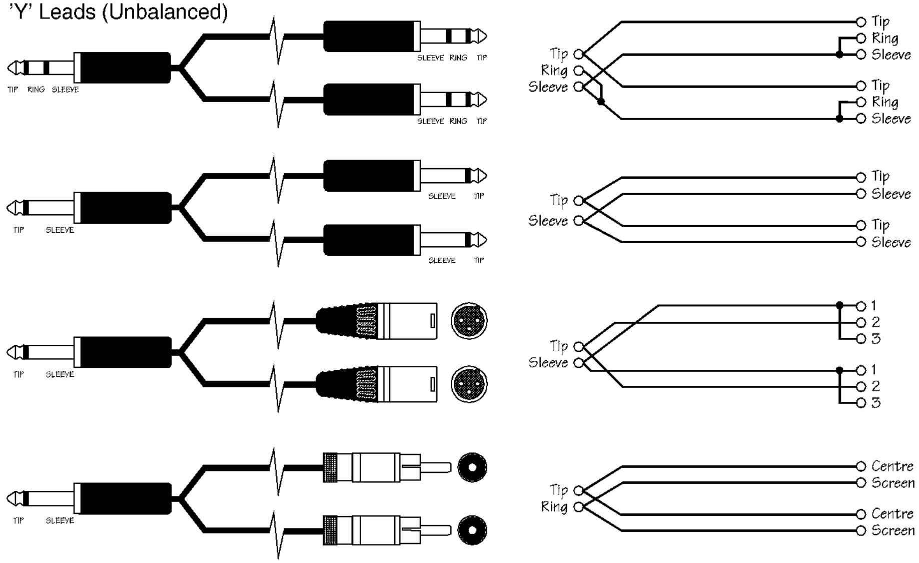
Where used....Aux, Mix outputs


SLEEVE RING TIP
TIP RING SLEEVE
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
Tip
Ring
Sleeve
1
2
3
1
2
3
1
2
3
TIP RING SLEEVE
Tip
Ring
Sleeve
1
2
3
1
2
3

Headphone Splitter


TIP RING SLEEVE
Tip
Ring
Sleeve
Tip
Ring
Sleeve
Tip
Ring
Sleeve

'Y' Leads (Unbalanced)


SLEEVE RING TIP
TIP RING SLEEVE
SLEEVE RING TIP
Tip
Ring
Sleeve
Tip
Ring
Sleeve
Tip
Ring
Sleeve
SLEEVE TIP
TIP SLEEVE
SLEEVE TIP
Tip
Sleeve
Tip
Sleeve
Tip
Sleeve
TIP SLEEVE
Tip
Sleeve
1
2
3
1
2
3
TIP SLEEVE
Tip
Ring
Centre
Screen
Centre
Screen

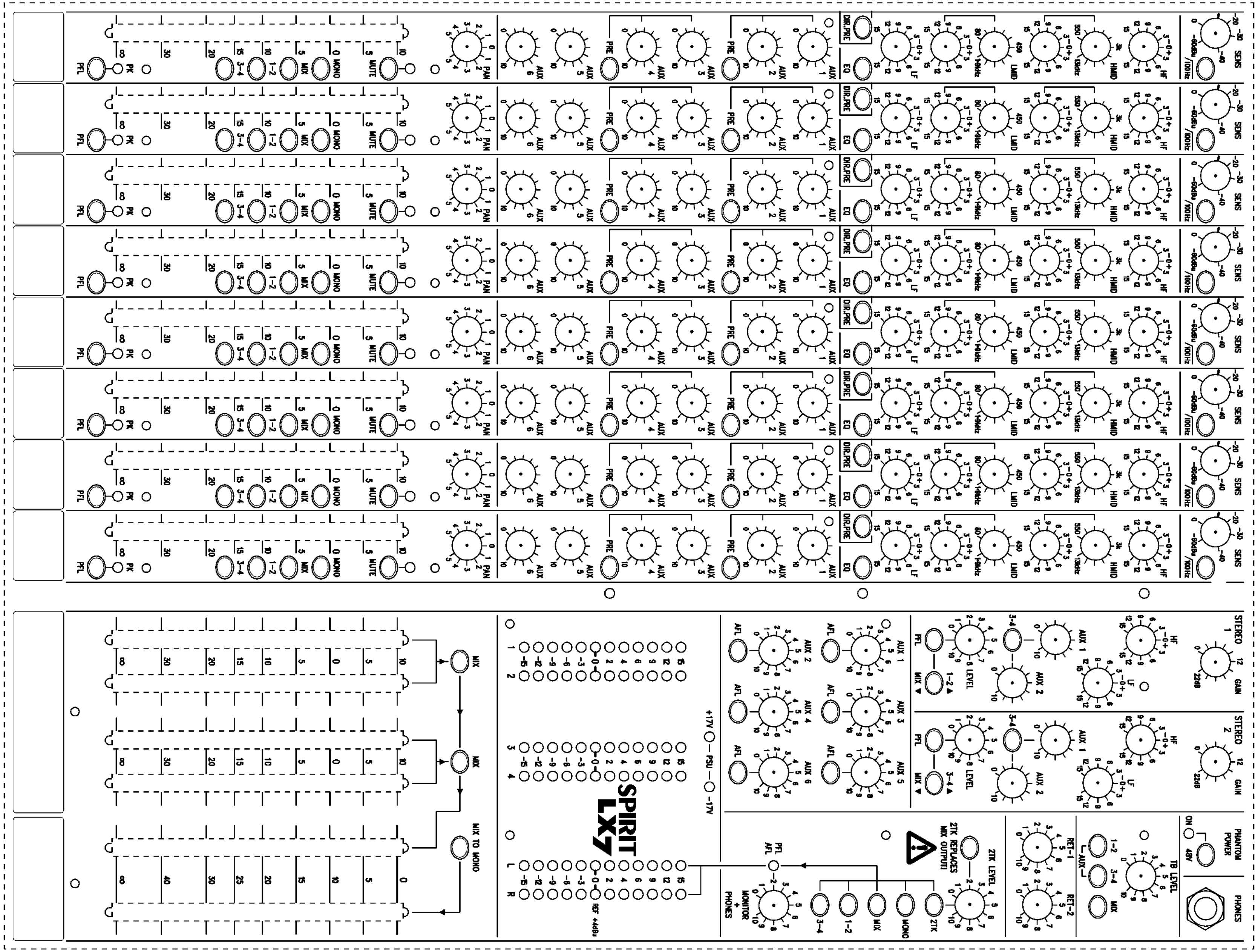
Control Position Sheet

You may freely copy this page to mark control positions to assist in resetting the desk between performances.

Gig :

Engineer :

Date :



SPIRIT
LX7
SENS
-20 -30 -40
-60dBu
/100Hz
HF
HMID
3k
13kHz
550
LMID
450
1.9kHz
80
LF
DIR.PRE
EQ
AUX 1
AUX 2
AUX 3
AUX 4
AUX 5
AUX 6
PRE
PAN
MUTE
MONO
MIX
1-2
3-4
PK
PFL
MIX TO MONO
+17V
PSU
-17V
REF +4dBu
L
R
AFL
STEREO 1
STEREO 2
GAIN
22dB
LEVEL
MIX ▼
1-2 ▲
3-4 ▲
PHANTOM POWER
ON
48V
PHONES
TB LEVEL
RET-1
RET-2
2TK LEVEL
2TK REPLACES MIX OUTPUT!
2TK
MONITOR + PHONES

Sheet Title:

Sheet Title:

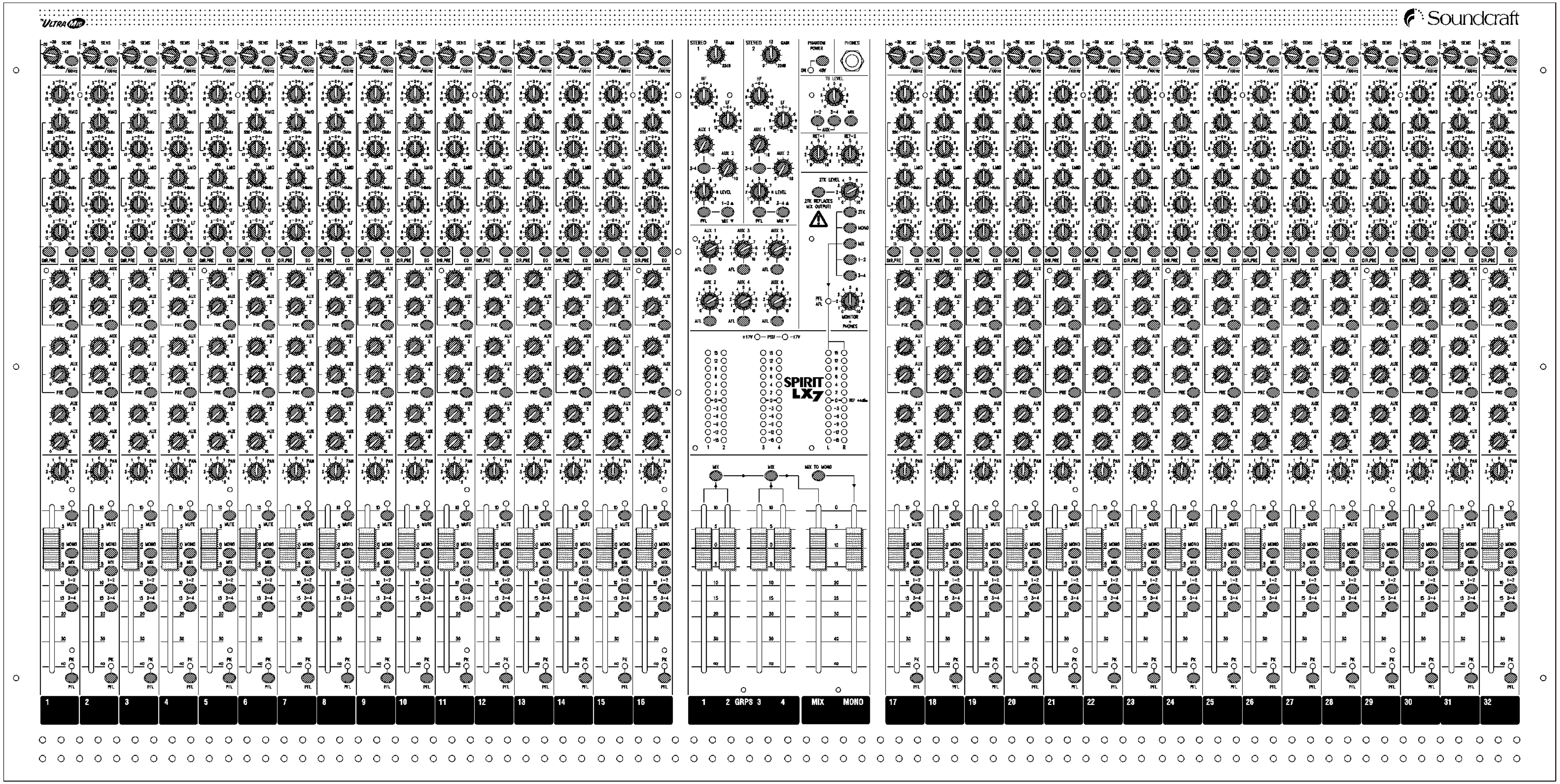
Sheet Title:

Sheet Title:

Sheet Title:

Sheet Title:

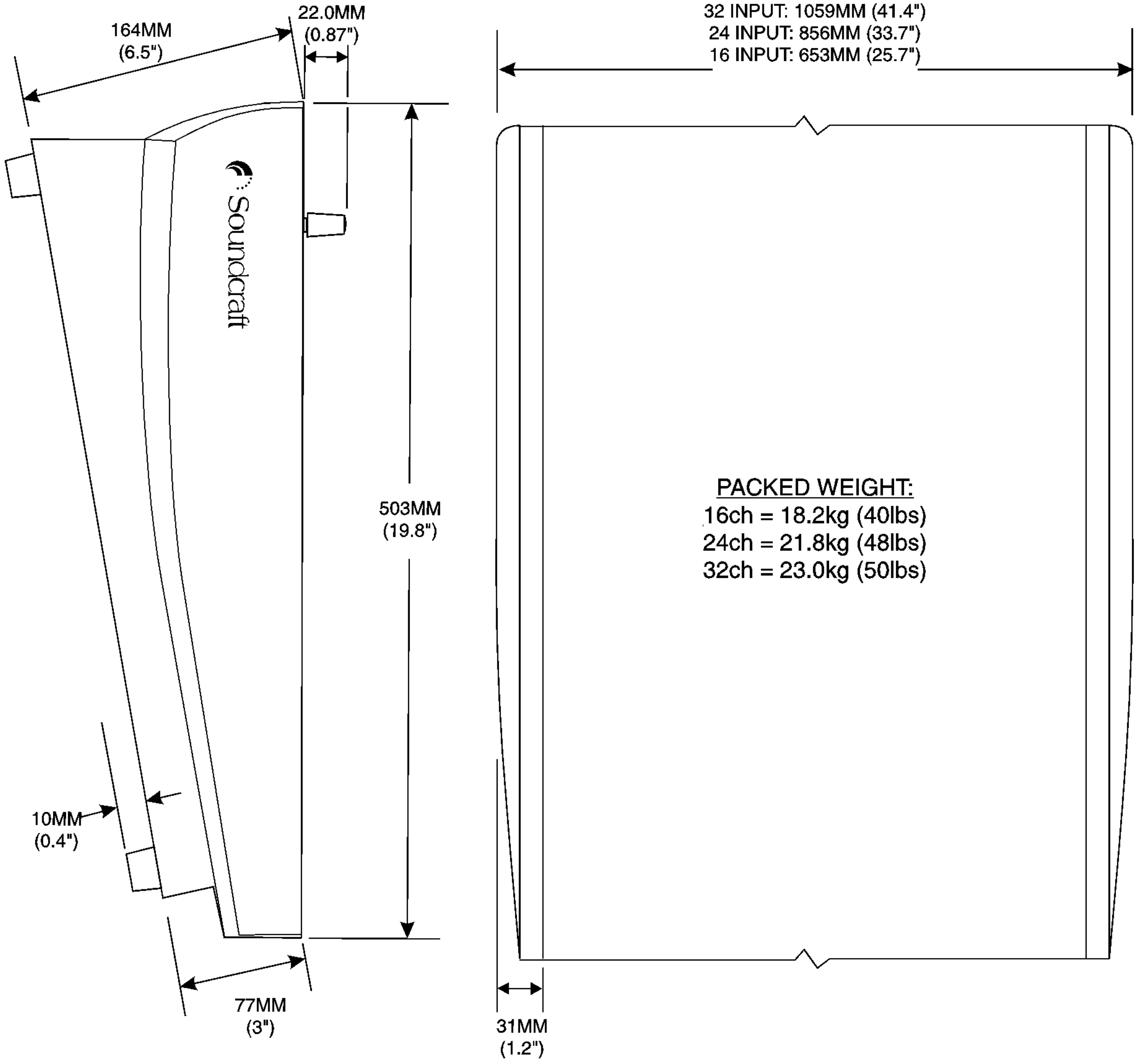
Sheet Title:


ULTRA MIC
Soundcraft
SPIRIT LX7
STEREO 1
STEREO 2
PHANTOM POWER
PHONES
2TK LEVEL
2TK REPLACES MIX OUTPUT
MONITOR PHONES
+17V PSU -17V
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16
1 2 GRPS 3 4
MIX
MONO
17
18
19
20
21
22
23
24
25
26
27
28
29
30
31
32

LX7 Console - Typical Starting Out Control Positions.

Dimensions

Abmessungen
Dimensions
Dimensioni
Dimensiones



PACKED WEIGHT:
16ch = 18.2kg (40lbs)
24ch = 21.8kg (48lbs)
32ch = 23.0kg (50lbs)

Specifications

Technische Daten
Specifications
Specifiche
Especificaciones

Noise

Measured RMS, 22Hz to 22kHz Bandwidth

Mic E.I.N. @ unity gain,150Ω source impedance	-129dBu
Mix Output, 24 inputs routed to mix, faders at unity, muted	<-80dBu
Mix Output, 24 inputs routed to mix, faders down	<-100dBu
Input to Direct output at unity gain	<-90dBu
Input to Direct output at 40dB gain	<-80dBu

Crosstalk (@1kHz, typical)

Fader attenuation	>95dB
Aux attenuation	>80dB
Pan isolation	>75dB
Adjacent channel crosstalk	>-80dB
Channel Mute	>90dB
Fader cut-off (rel. 0 mark)	>90dB
Aux Send pots offness	>80dB

Frequency Response

Mic/Line Input to any output, 20Hz - 20kHz	<1dB

THD + N

Mic sens. -30dBu, +20dBu at all outputs @1kHz	<0.006%

CMRR

Typical @ max gain @ 1kHz	>80dB
Typical @ any gain @ 50Hz	>60dB

Input & Output Impedance

Mic Input	1.8kΩ
Line Input	10kΩ
Stereo Input	8.6kΩ
2TK Return	12kΩ
Mix, Aux, Direct Outputs, and Insert Sends	75Ω

Input & Output Levels

Mic Input max. level	+22dBu
Line Input max. level	+22dBu
Stereo Input max. level	+22dBu
2TK Return	>30dBu
Headphones (@ 200Ω)	150mW

Weight

Exc. power supply unit	16/2	3.4 kg/7.5 lb
	14/2	3.0 kg/6.6 lb

Soundcraft
Harman International Industries Ltd., Cranborne House
Cranborne Road, Potters Bar, Hertfordshire., EN6 3JN U.K.
Tel: +44 (0) 1707 665000 Fax: +44 (0) 1707 660742
http://www.spiritbysoundcraft.com
http://www.soundcraft.com

A Harman International Company

Soundcraft USA
Air Park Business Center 12
1449 Donelson Pike
Nashville, TN 37217, USA
Tel: +1 (615) 360 0707 Fax: +1 (615) 360 0273
Toll Free within USA only: 800 255 4363

This equipment complies
with the EMC Directive
89/336/EEC
and LVD 73/23/EEC
and 93/68/EEC

For further details contact the address on the left

Issue 3 Part No: ZM0188

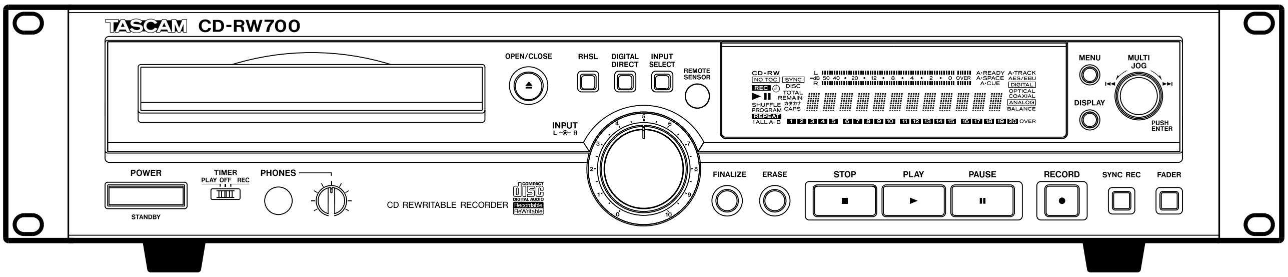
3D0031900A

TASCAM
TEAC Professional Division

CD-RW700

CD Rewritable Recorder

COMPACT disc DIGITAL AUDIO
Recordable
ReWritable


TASCAM CD-RW700
OPEN/CLOSE
RHSL
DIGITAL DIRECT
INPUT SELECT
REMOTE SENSOR
INPUT L R
MENU
MULTI JOG
DISPLAY
PUSH ENTER
POWER
STANDBY
TIMER PLAY OFF REC
PHONES
CD REWRITABLE RECORDER
FINALIZE
ERASE
STOP
PLAY
PAUSE
RECORD
SYNC REC
FADER

OWNER'S MANUAL

CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK). NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock to persons..

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the appliance.

This appliance has a serial number located on the rear panel. Please record the model number and serial number and retain them for your records.

Model number____________________

Serial number ____________________

WARNING: TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

Note for U.K. Customers

DO NOT cut off the mains plug from this equipment.
If the plug fitted is not suitable for the power points in your home or the cable is too short to reach a power point, then obtain an appropriate safety approved extension lead or consult your dealer.

If nonetheless the mains plug is cut off, remove the fuse and dispose of the plug immediately, to avoid a possible shock hazard by inadvertent connection to the mains supply.

If this product is not provided with a mains plug, or one has to be fitted, then follow the instructions given below:

IMPORTANT: DO NOT make any connection to the larger terminal which is marked with the letter E or by the safety earth symbol ⏚ or coloured GREEN or GREEN-and-YELLOW.

The wires in the mains lead on this product are coloured in accordance with the following code:

BLUE : NEUTRAL
BROWN : LIVE

As these colours may not correspond with the coloured markings identifying the terminals in your plug proceed as follows:

The wire which is coloured BLUE must be connected to the terminal which is marked with the letter N or coloured BLACK.

The wire which is coloured BROWN must be connected to the terminal which is marked with the letter L or coloured RED.

When replacing the fuse only a correctly rated approved type should be used and be sure to re-fit the fuse cover.

IF IN DOUBT — CONSULT A COMPETENT ELECTRICIAN.

For U.S.A

TO THE USER

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against interference in a residential area. This device generates and uses radio frequency energy and if not installed and used in accordance with the instructions, it may cause interference to radio or TV reception. If this unit does cause interference with TV or radio reception you can try to correct the interference by one or more of the following measures:

a) Reorient or relocate the receiving antenna.
b) Increase the separation between the equipment and the receiver.
c) Plug the equipment into a different outlet so that it is not on the same circuit as the receiver.
If necessary, consult the dealer or an experienced radio/TV technician for additional suggestions.

CAUTION
Changes or modifications to this equipment not expressly approved by TEAC CORPORATION for compliance could void the user’s authority to operate

For the consumers in Europe

WARNING
This is a Class A product. In a domestic environment, this product may cause radio interference in which case the user may be required to take adequate measures.

Pour les utilisateurs en Europe

AVERTISSEMENT
Il s’agit d’un produit de Classe A. Dans un environnement domestique, cet appareil peut provoquer des interférences radio, dans ce cas l’utilisateur peut être amené à prendre des mesures appropriées.

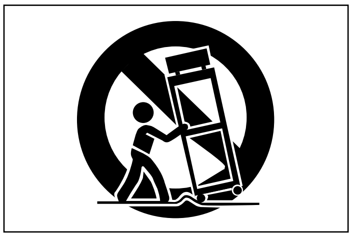
Für Kunden in Europa

Warnung
Dies is eine Einrichtung, welche die Funk-Entstörung nach Klasse A besitzt. Diese Einrichtung kann im Wohnbereich Funkstörungen versursachen ; in diesem Fall kann vom Betrieber verlang werden, angemessene Maßnahmen durchzuführen und dafür aufzukommen.

IMPORTANT SAFETY INSTRUCTIONS

CAUTION:
- **Read all of these Instructions.**
- **Save these Instructions for later use.**
- **Follow all Warnings and Instructions marked on the audio equipment.**

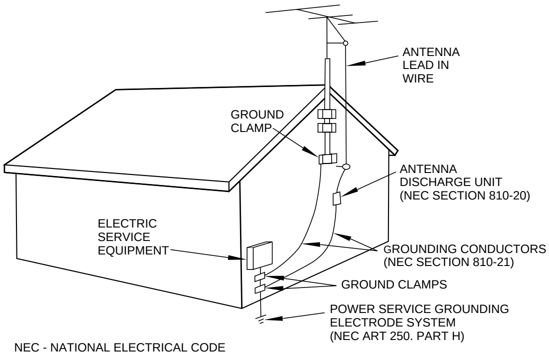
1) Read Instructions — All the safety and operating instructions should be read before the product is operated.
2) Retain Instructions — The safety and operating instructions should be retained for future reference.
3) Heed Warnings — All warnings on the product and in the operating instructions should be adhered to.
4) Follow Instructions — All operating and use instructions should be followed.
5) Cleaning — Unplug this product from the wall outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a damp cloth for cleaning.
6) Attachments — Do not use attachments not recommended by the product manufacturer as they may cause hazards.
7) Water and Moisture — Do not use this product near water — for example, near a bath tub, wash bowl, kitchen sink, or laundry tub; in a wet basement; or near a swimming pool; and the like.
8) Accessories — Do not place this product on an unstable cart, stand, tripod, bracket, or table. The product may fall, causing serious injury to a child or adult, and serious damage to the product. Use only with a cart, stand, tripod, bracket, or table recommended by the manufacturer, or sold with the product. Any mounting of the product should follow the manufacturer's instructions, and should use a mounting accessory recommended by the manufacturer.
9) A product and cart combination should be moved with care. Quick stops, excessive force, and uneven surfaces may cause the product and cart combination to overturn.



10) Ventilation — Slots and openings in the cabinet are provided for ventilation and to ensure reliable operation of the product and to protect it from overheating, and these openings must not be blocked or covered. The openings should never be blocked by placing the product on a bed, sofa, rug, or other similar surface. This product should not be placed in a built-in installation such as a bookcase or rack unless proper ventilation is provided or the manufacturer's instructions have been adhered to.
11) Power Sources — This product should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power supply to your home, consult your product dealer or local power company. For products intended to operate from battery power, or other sources, refer to the operating instructions.
12) Grounding or Polarization — This product may be equipped with a polarized alternating-current line plug (a plug having one blade wider than the other). This plug will fit into the power outlet only one way. This is a safety feature. If you are unable to insert the plug fully into the outlet, try reversing the plug. If the plug should still fail to fit, contact your electrician to replace your obsolete outlet. Do not defeat the safety purpose of the polarized plug.
13) Power-Cord Protection — Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the product.
14) Outdoor Antenna Grounding — If an outside antenna or cable system is connected to the product, be sure the antenna or cable system is grounded so as to provide some protection against voltage surges and built-up static charges. Article 810 of the National Electrical Code, ANSI/NFPA 70, provides information with regard to proper grounding of the mast and supporting structure, grounding of the lead-in wire to an antenna discharge unit, size of grounding conductors, location of antenna-discharge unit, connection to grounding electrodes, and requirements for the grounding electrode.

"Note to CATV system installer:
This reminder is provided to call the CATV system installer's attention to Section 820-40 of the NEC which provides guidelines for proper grounding and, in particular, specifies that the cable ground shall be connected to the grounding system of the building, as close to the point of cable entry as practical.

Example of Antenna Grounding as per National Electrical Code, ANSI/NFPA 70


ANTENNA LEAD IN WIRE
GROUND CLAMP
ANTENNA DISCHARGE UNIT (NEC SECTION 810-20)
ELECTRIC SERVICE EQUIPMENT
GROUNDING CONDUCTORS (NEC SECTION 810-21)
GROUND CLAMPS
POWER SERVICE GROUNDING ELECTRODE SYSTEM (NEC ART 250. PART H)
NEC - NATIONAL ELECTRICAL CODE

15) Lightning — For added protection for this product during a lightning storm, or when it is left unattended and unused for long periods of time, unplug it from the wall outlet and disconnect the antenna or cable system. This will prevent damage to the product due to lightning and power-line surges.
16) Power Lines — An outside antenna system should not be located in the vicinity of overhead power lines or other electric light or power circuits, or where it can fall into such power lines or circuits. When installing an outside antenna system, extreme care should be taken to keep from touching such power lines or circuits as contact with them might be fatal.
17) Overloading — Do not overload wall outlets, extension cords, or integral convenience receptacles as this can result in risk of fire or electric shock.
18) Object and Liquid Entry — Never push objects of any kind into this product through openings as they may touch dangerous voltage points or short-out parts that could result in a fire or electric shock. Never spill liquid of any kind on the product.
19) Servicing — Do not attempt to service this product yourself as opening or removing covers may expose you to dangerous voltage or other hazards. Refer all servicing to qualified service personnel.
20) Damage Requiring Service — Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:
a) when the power-supply cord or plug is damaged.
b) if liquid has been spilled, or objects have fallen into the product.
c) if the product has been exposed to rain or water.
d) if the product does not operate normally by following the operating instructions. Adjust only those controls that are covered by the operating instructions as an improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the product to its normal operation.
e) if the product has been dropped or damaged in any way.
f) when the product exhibits a distinct change in performance – this indicates a need for service.
21) Replacement Parts — When replacement parts are required, be sure the service technician has used replacement parts specified by the manufacturer or have the same characteristics as the original part.
Unauthorized substitutions may result in fire, electric shock, or other hazards.
22) Safety Check — Upon completion of any service or repairs to this product, ask the service technician to perform safety checks to determine that the product is in proper operating condition.
23) Wall or Ceiling Mounting — The product should be mounted to a wall or ceiling only as recommended by the manufacturer.
24) Heat — The product should be situated away from heat sources such as radiators, heat registers, stoves, or other products (including amplifiers) that produce heat.

SAFETY INFORMATION

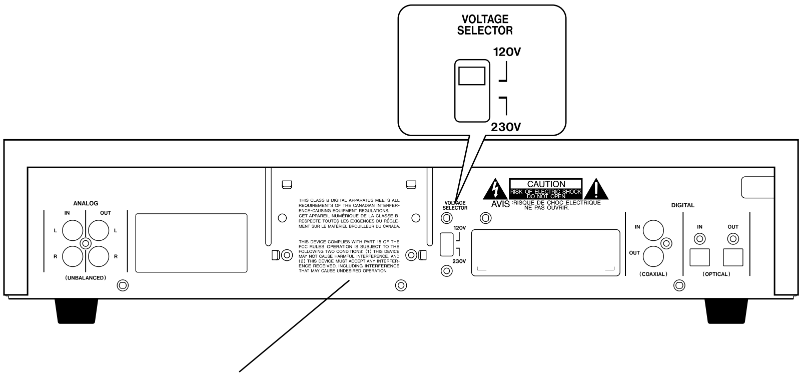
This product has been designed and manufactured according to FDA regulations "title 21, CFR, chapter 1, subchapter J, based on the Radiation Control for Health and Safety Act of 1968", and is classified as a class 1 laser product. There is no hazardous invisible laser radiation during operation because invisible laser radiation emitted inside of this product is completely confined in the protective housings.
The label required in this regulation is shown ①.

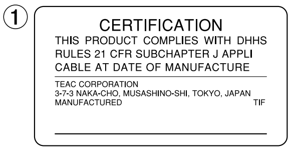
CAUTION

- DO NOT REMOVE THE PROTECTIVE HOUSING USING A SCREWDRIVER.
- USE OF CONTROLS OR ADJUSTMENTS OR PERFORMANCE OF PROCEDURES OTHER THAN THOSE SPECIFIED HEREIN MAY RESULT IN HAZARDOUS RADIATION EXPOSURE.
- IF THIS PRODUCT DEVELOPS TROUBLE, CONTACT YOUR NEAREST QUALIFIED SERVICE PERSONNEL, AND DO NOT USE THE PRODUCT IN ITS DAMAGED STATE.

Voltage Conversion
(General export models only)
Be sure to remove the power cord from the AC outlet before repositioning the voltage converter switch.

1.Locate the voltage selector on the rear panel.
2.Using a flat-bladed screwdriver, set to the appropriate 230V or 120V position according to your area.


VOLTAGE SELECTOR
120V
230V
ANALOG
IN
OUT
L
R
(UNBALANCED)
THIS CLASS B DIGITAL APPARATUS MEETS ALL REQUIREMENTS OF THE CANADIAN INTERFERENCE-CAUSING EQUIPMENT REGULATIONS.
CET APPAREIL NUMÉRIQUE DE LA CLASSE B RESPECTE TOUTES LES EXIGENCES DU RÉGLEMENT SUR LE MATÉRIEL BROUILLEUR DU CANADA.
THIS DEVICE COMPLIES WITH PART 15 OF THE FCC RULES. OPERATION IS SUBJECT TO THE FOLLOWING TWO CONDITIONS: (1) THIS DEVICE MAY NOT CAUSE HARMFUL INTERFERENCE, AND (2) THIS DEVICE MUST ACCEPT ANY INTERFERENCE RECEIVED, INCLUDING INTERFERENCE THAT MAY CAUSE UNDESIRED OPERATION.
VOLTAGE SELECTOR
120V
230V
CAUTION
RISK OF ELECTRIC SHOCK
DO NOT OPEN
AVIS :RISQUE DE CHOC ELECTRIQUE
NE PAS OUVRIR.
DIGITAL
IN
OUT
(COAXIAL)
(OPTICAL)

①

CERTIFICATION
THIS PRODUCT COMPLIES WITH DHHS RULES 21 CFR SUBCHAPTER J APPLICABLE AT DATE OF MANUFACTURE
TEAC CORPORATION
3-7-3 NAKA-CHO, MUSASHINO-SHI, TOKYO, JAPAN
MANUFACTURED TIF

For U.S.A.

Optical pickup:		
Type	:	KRS-202A or KRS-220B
Manufacturer	:	SONY Corporation
Laser output	:	Less than 0.1 mW (Play) and 32 mW (Record) on the objective lens
Wavelength	:	777 - 787 nm

For CANADA

AC POWER CORD CONNECTION
CAUTION:
TO PREVENT ELECTRIC SHOCK, MATCH WIDE BLADE OF PLUG TO WIDE SLOT, FULLY INSERT.

Pour le CANADA

CORDE DE CONNEXION CA
ATTENTION:
POUR ÉVITER LES CHOCS ÉLECTRIQUES, INTRODUIRE LA LAME LA PLUS LARGE DE LA FICHE DANS LA BORNE CORRESPONDANTE DE LA PRISE ET POUSSER JUSQU’AU FOND.

Table of Contents

1 - Introduction

1.1 About CD-R and CD-RW discs 6
1.2 Finalizing 7
1.2.1 “Recordable” discs 7
1.2.2 About this manual 7
1.3 Unpacking the CD-RW700 7
1.3.1 Rack-mounting the unit 7
1.4 Environmental considerations 7
1.4.1 Beware of condensation 8
1.4.2 Never use a stabilizer or printable discs . 8
1.4.3 Handling of compact discs 8
1.4.4 Additional notes with regard to CD-R and CD-RW discs 8
1.4.5 Recommended media 9
1.4.6 Default settings 9
1.4.7 Dimensional drawing 9

2 - Features of the CD-RW700

2.1 Front panel 11
① POWER switch (STANDBY/ON) 11
② Disc tray 11
③ OPEN/CLOSE key 11
④ RHSL key 11
⑤ DIGITAL DIRECT 11
⑥ INPUT SELECT 11
⑦ REMOTE SENSOR 11
⑧ Display 11
⑨ MENU key 11
⑩ DISPLAY key 11
⑪ MULTI JOG control 11
⑫ TIMER switch (PLAY, OFF, REC) 11
⑬ PHONES jack and LEVEL control 11
⑭ REC LEVEL controls 11
⑮ FINALIZE key 11
⑯ ERASE key 11
⑰ STOP key 12
⑱ PLAY key 12
⑲ PAUSE key 12
⑳ RECORD key 12
㉑ SYNC REC key 12
㉒ FADER key 12
2.2 Rear panel 12
㉓ ANALOG IN (L, R) (UNBALANCED) . 12
㉔ ANALOG OUT (L, R) (UNBALANCED) 12
㉕ DIGITAL COAXIAL (IN, OUT) 12
㉖ DIGITAL OPTICAL (IN, OUT) 12
2.3 Remote control unit 12
❶ POWER key 12
❷ OPEN/CLOSE key 12
❸ Number keys (0 through 10 and +10) 12
❹ CLEAR key 12
❺ REPEAT key 12
❻ DISPLAY key 12
❼ MENU key 12
❽ ENTER key 12
❾ FADER key 12
❿ MONITOR key 12
⓫ PLAY MODE key 12
⓬ FINALIZE key 12
⓭ ERASE key 12
⓮ RECORD key 12
⓯ SYNC REC key 13
⓰ REC MUTE key 13
⓱ SEARCH keys 13
⓲ SKIP keys 13
⓳ STOP key 13
⓴ PAUSE key 13
㉑ PLAY key 13
2.3.1 Use of the remote control unit 13
2.4 Connections 13
2.5 Timer operations 13
2.5.1 Setting the clock 13
2.5.2 Setting the timer 14
2.5.3 Using the timer 14

3 - Playback operations

3.1 Simple playback operations 15
3.1.1 Time display modes 15
3.1.2 Track search 15
3.1.3 Playback modes 15
3.2 Programmed order playback 16
3.3 Repeat play 16

4 - Recording

4.1 Input selection 17
4.1.1 Signal volume 17
4.1.2 Input monitoring 17
4.1.3 Frequency conversion 17
4.2 Basic recording 18
4.2.1 Playing back tracks 18
4.2.2 Time display while recording 18
4.2.3 Manual track division 19
4.2.4 Restricting the number of tracks 19
4.2.5 Full disc 19
4.3 Advanced recording 19
4.3.1 Synchronized recording 19
4.3.2 Automatic track division 21
4.3.3 SCMS and the CD-RW700 22
4.3.4 Digital direct 22
4.3.5 Rec mute function 23
4.3.6 Fade-in and fade-out 23
4.4 Rehearsal 23
4.4.1 Rehearsal for synchronized recording .. 24
4.4.2 Rehearsal for track division 25

5 - After recording

5.1 Finalizing 26
5.1.1 Unfinalizing CD-RW discs 26
5.2 Erasing 26
5.2.1 Erasing tracks 26
5.2.2 Erasing a whole disc 27
5.2.3 Refreshing a disc 27

6 - Reference and specifications

6.1 Troubleshooting 28
6.2 Specifications 28

The CD-RW700 CD rewritable recorder allows you to make your own CDs from a variety of sources—other CDs, MD recordings, analog cassette tapes, and records.

Tracks can be recorded all at one time, or added, until the disc is “finalized”, allowing it to be played on CD players.

NOTE

Some models of CD player may be unable to play certain discs produced using the CD-RW700.

A variety of sophisticated features are incorporated in the CD-RW700, allowing a wide degree of flexibility.

The CD-RW700 provides analog unbalanced audio inputs and outputs for connection to –10 dBV equipment.

In addition, coaxial and optical inputs and outputs are provided for recording from and playing back to suitably-equipped digital audio equipment.

The analog and digital signals may be mixed, allowing unprecedented versatility while recording.

24-bit digital-to-analog and analog-to-digital converters provide excellent recording and playback quality.

A sampling frequency convertor is incorporated, allowing the recording of CDs from digital audio sources which are at frequencies other than the CD standard sampling frequency of 44.1 kHz.

Track divisions can be entered manually, or can produced automatically as a response to the input signal level.

A recording buffer ensures that even the first milliseconds of a track are recorded, and a rehearsal mode allows the precise positioning of track divisions, allowing the elimination of false starts or late entries.

“One-touch” fade-ins and fade-outs over a specified time allow you to edit program material to your specific requirements.

A multi-function “multi jog” dial is used to set and confirm parameter settings.

A wireless remote control unit allows control of the CD-RW700 from remote locations.

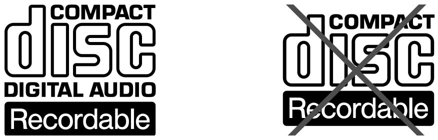
NOTE

This product is designed to help you record and reproduce sound works to which you own the copyright, or where you have obtained permission from the copyright holder or the rightful licensor. Unless you own the copyright, or have obtained the appropriate permission from the copyright holder or the rightful licensor, your unauthorized recording, reproduction or distribution thereof may result in severe criminal penalties under copyright laws and international copyright treaties. If you are uncertain about your rights, contact your legal advisor. Under no circumstances will TEAC Corporation be responsible for the consequences of any illegal copying performed using the CD-RW700.

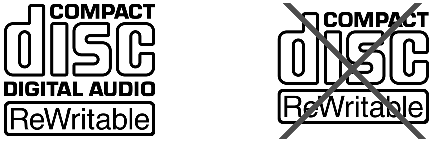
1.1 About CD-R and CD-RW discs

The CD-RW700 uses CD-R and CD-RW Digital media.

CD-R discs can be recorded once only. Once they have been used for recording, they cannot be erased or re-recorded. However, if space is available on the disc, additional material can be recorded. The packaging of CD-R discs will include the following logo:



By contrast, a CD-RW disc can be used in much the same way as a CD-R disc, but the last track or tracks recorded can be erased before the disc has been “finalized”, and the space on the disc can be re- used for other recordings. The packaging of CD-RW discs will include the a logo similar to the following:



However, you should note that an audio CD created using a CD-RW disc may not play back satisfactorily on every audio CD player. It will, naturally, be playable on the CD-RW700. This is in no way a limitation of the CD-RW700 itself, but of the difference between the different types of media and the methods used to read them.

CD-R discs created on the CD-RW700, by contrast, may be played satisfactorily on the majority of audio CD players.

For details of disc brands that are known to work well with the CD-RW700, see 1.4.5, “Recommended media”.

1.2 Finalizing

Although audio data may be written on a CD-R or CD-RW disc, a standard CD player will not be able to read the data (i.e. play back the audio) until a final table of contents (TOC) has been written at the start of the disc.

The process of writing this table of contents is known as “finalizing”. Once this has been done, no further data can be written to the disk. See 5.1, “Finalizing” for details.

In the case of a CD-RW disc which has been finalized, the whole of the disc may be erased, or the disc may be “refreshed”, and the disc re-used.

In addition, a CD-RW disc that has been finalized may be “unfinalized”, i.e. the TOC is removed. This allows further tracks to be recorded to the disc, provided that there is space on the disc.

1.2.1 “Recordable” discs

In this manual, we use the term “recordable” disc to describe a CD-R or CD-RW disc that has not been finalized, i.e. further recording is possible on the disc.

1.2.2 About this manual

In this manual, we use the following conventions:

- The names of keys and controls are given in the following typeface: **ERASE**.
- When the alphanumeric portion of the display shows a message, this is shown in the following typeface: `Welcome`.
- If a preset indicator in the display (i.e. one which cannot change, but is either on or off) is shown, this is shown as follows: SHUFFLE.
- When referring to a front or rear panel control or feature, the following typeface is used to show the number of the feature (as given in the diagram in this manual, and the corresponding explanation): ⑤.
- When referring to a remote control key, the following typeface is used to show the number of the key (as given in the diagram in this manual, and the corresponding explanation): ❺.

1.3 Unpacking the CD-RW700

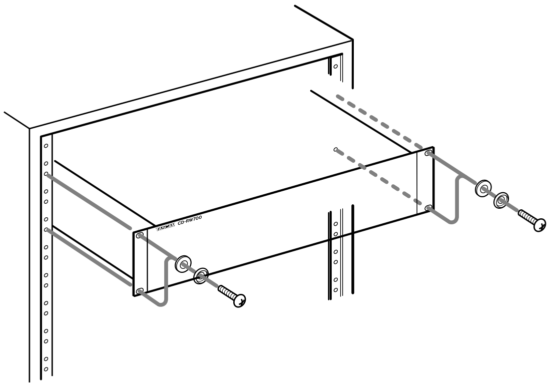
When unpacking the unit, you should find the following items, in addition to this manual:

- The main unit itself
- The remote control unit
- Two AA batteries for the remote control unit
- A rack-mounting screw kit

1.3.1 Rack-mounting the unit

Use the rack-mounting kit to mount the unit in a standard 19-inch equipment rack, as shown below. Remove the feet of the unit before mounting it.

See 1.4, “Environmental considerations” below for details of ventilation, etc.



1.4 Environmental considerations

The CD-RW700 may be used in most areas, but to maintain top performance, and prolong operating life, observe the following environmental conditions:

The nominal temperature should be between 5˚C and 35˚C (41˚F and 95˚F). The CD-RW700 is more sensitive to extremes of temperature than ordinary CD players.

Relative humidity should be 30 to 90 degrees noncondensing.

As the unit may become hot during operation, always leave sufficient space above the unit for ventilation. If you are mounting the unit in a rack, leave 1U of space above it.

You should not place the unit on a piece of equipment generating heat, e.g. an amplifier, to avoid possible problems with overheating.

Make sure that the unit is mounted in a level position for correct operation. Do not mount the unit in a rack tilted 5˚ or more from the vertical position.

NOTE

If the mounting surface is more than 5 degrees away from the horizontal, it will not open or close.

The voltage supplied to the unit should match the voltage as printed on the rear panel. If you are in any doubt regarding this matter, consult an electrician.

NOTE

When transporting the unit, always use the original packing materials. For this reason, we strongly recommend that you save all the packing materials that came with the CD-RW700, in case you need to transport it in the future.

In addition, when transporting the unit, you should tape the disc tray closed, using an adhesive tape that will not spoil the finish of the unit.

1.4.1 Beware of condensation

If the unit (or a compact disc) is moved from a cold to a warm place, or used after a sudden temperature change, there is a danger of condensation; vapor in the air could condense on the internal mechanism, making correct operation impossible. To prevent this, or if this occurs, leave the player for one or two hours with the power turned on, then turn the power off and on again.

1.4.2 Never use a stabilizer or printable discs

Using commercially available CD stabilizers or printable recordable discs with this player will damage the mechanism and cause it to malfunction.

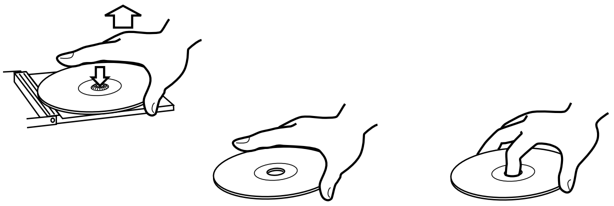
NOTE

Never use a disc that has had a stabilizer mounted to it. The residual adhesive may cause the disc to stick to the mechanism of the CD-RW700. If it sticks to the mechanism, you will need a technician to get it out.

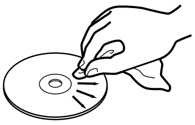
1.4.3 Handling of compact discs

Observe the following:

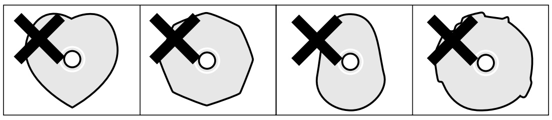
- Always place the compact discs in the trays with their label facing upward (compact discs can only be played on one side).
- To remove a disc from its storage case, press down on the center of the case, and lift the disc out, holding it carefully by the edges.



- Finger marks and dust should be carefully wiped off the disc's recorded surface with a soft cloth. Unlike conventional records, the compact disc has no grooves to collect dust and microscopic debris, so gently wiping with a soft cloth should remove most particles.
- Wipe in a straight motion from the inside to the outside of the disc. Small dust particles and light stains will have absolutely no effect on reproduction quality.



- Never use such chemicals as record sprays, anti-static sprays or fluid, benzine or thinner to clean compact discs. Such chemicals will do irreparable damage to the disc's plastic surface.
- Discs should be returned to their cases after use to avoid serious scratches that could cause the laser pickup to "skip".
- Don't expose discs to direct sunlight or high humidity and temperature for extended periods. Long exposure to high temperature can warp the disc.
- Only use circular compact discs. Avoid using non-circular promotional, etc. discs.



- To keep the laser pickups clean, don't touch them. For the same reason, don't leave the disc trays opened unnecessarily.

1.4.4 Additional notes with regard to CD-R and CD-RW discs

There are additional precautions that you should take when handling CD-R and CD-RW discs, that are different to those that you should take when handling ordinary CDs.

- Avoid touching the recording (non-label) side of a disc on which you will be recording. Recording on a disc requires a cleaner surface than playing back, and fingerprints, grease, etc. can cause errors in the recording process.
- CD-R discs are more sensitive to the effects of heat and ultraviolet rays than ordinary CDs. It is important that they are not stored in a location where

direct sunlight will fall on them, and which is away from sources of heat such as radiators or heat-generating electrical devices.

- Always store CD-R discs in their “jewel cases” to avoid dirt and dust accumulating on their surfaces.
- Do not put labels or protective sheets on the discs and do not use any protective coating spray.
- When labeling CD-R discs, always use a soft oil-based felt-tipped pen to write the information. Never use a ball-point or hard-tipped pen, as this may cause damage to the recorded side.
- Dirty discs may be cleaned using a soft dry cloth and/or a commercial CD cleaning fluid or ethyl alcohol. Do not use thinner, gasoline, benzene or LP cleaning fluid, as these will damage the disc.
- If you are in any doubt as to the care and handling of a CD-R disc, read the precautions supplied with the disc, or contact the disc manufacturer directly.

1.4.5 Recommended media

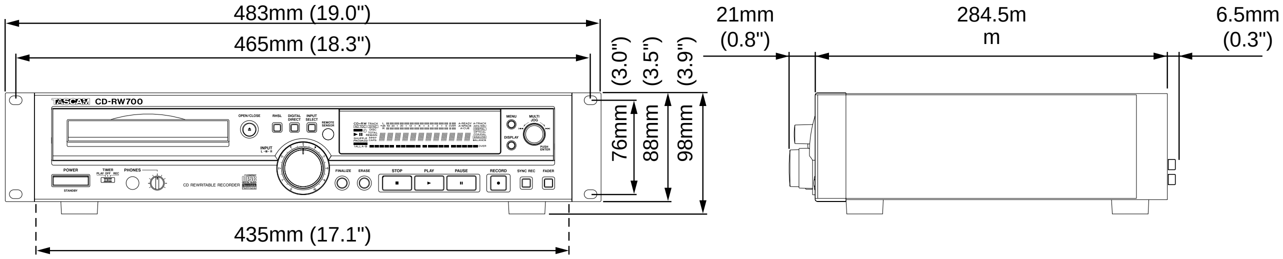
An up-to-date list of manufacturers who produce media suitable for use in the CD-RW700 can be obtained from your TASCAM dealer, or found on the TASCAM Web site at http://www.tascam.com.

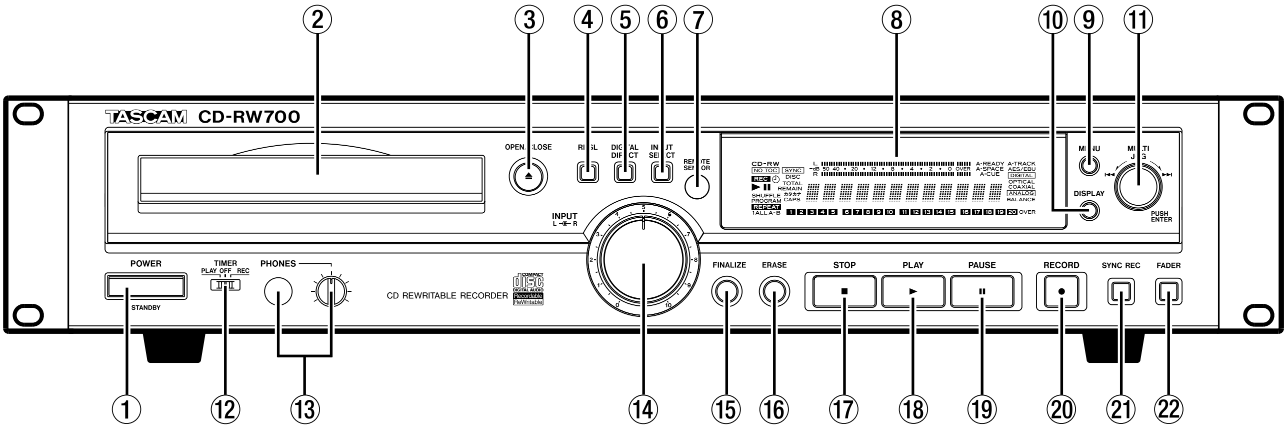
1.4.6 Default settings

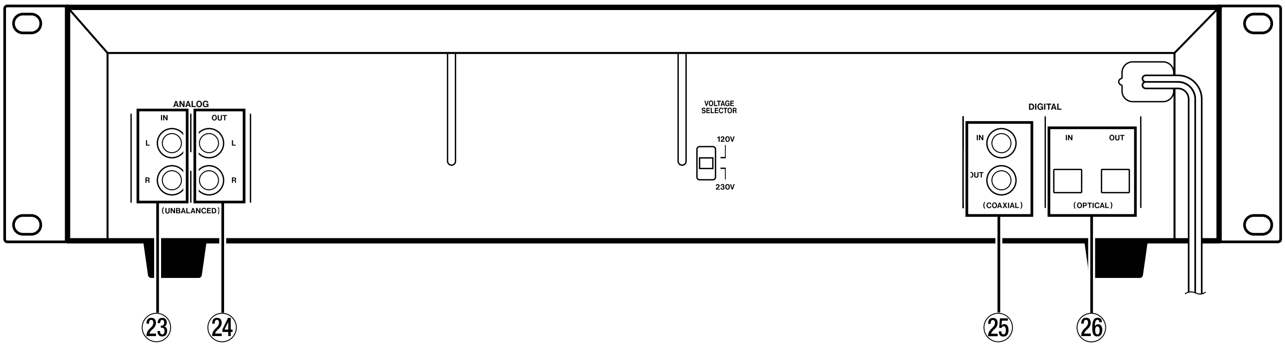
The default settings when the CD-RW700 is shipped from the factory are given below.

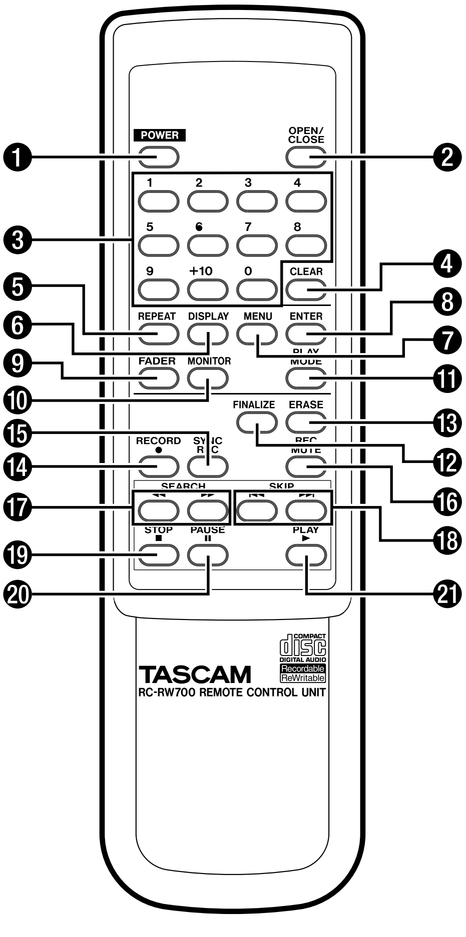
Setting	Description	Default value
Digital direct (D_DIRECT)	4.3.4, “Digital direct”	Off
Input selection	4.1, “Input selection”	Analog
Auto track (A_TRACK)	4.3.2, “Automatic track division”	On
Trigger level (S_LEVEL)	4.3.1, “Synchronized recording”	–54 dB
FADE IN time	4.3.6, “Fade-in and fade-out”	3 seconds
FADE OUT time	4.3.6, “Fade-in and fade-out”	3 seconds
Volume (DIGITAL IN)	4.1.1, “Signal volume”	0 dB
Volume (ANALOG IN)	4.1.1, “Signal volume”	0 dB
Timer on/off times	2.5.2, “Setting the timer”	Unset
Play mode	3.1.3, “Playback modes”	Continue
Trim	4.4, “Rehearsal”	±0 frames

1.4.7 Dimensional drawing




②
③
④
⑤
⑥
⑦
⑧
⑩
⑨
⑪
TASCAM CD-RW700
OPEN/CLOSE
RHSL
DIGITAL DIRECT
INPUT SELECT
REMOTE SENSOR
MENU
MULTI JOG
PUSH ENTER
DISPLAY
INPUT
L R
POWER
STANDBY
TIMER
PLAY OFF REC
PHONES
CD REWRITABLE RECORDER
FINALIZE
ERASE
STOP
PLAY
PAUSE
RECORD
SYNC REC
FADER
①
⑫
⑬
⑭
⑮
⑯
⑰
⑱
⑲
⑳
㉑
㉒


ANALOG
IN
OUT
L
R
(UNBALANCED)
VOLTAGE SELECTOR
120V
230V
DIGITAL
IN
OUT
(COAXIAL)
IN
OUT
(OPTICAL)
㉓
㉔
㉕
㉖


POWER
OPEN/CLOSE
1
2
3
4
5
6
7
8
9
+10
0
CLEAR
REPEAT
DISPLAY
MENU
ENTER
FADER
MONITOR
PLAY MODE
FINALIZE
ERASE
RECORD
SYNC REC
REC MUTE
SEARCH
SKIP
STOP
PAUSE
PLAY
TASCAM
RC-RW700 REMOTE CONTROL UNIT
❶
❷
❸
❹
❺
❻
❼
❽
❾
❿
⓫
⓬
⓭
⓮
⓯
⓰
⓱
⓲
⓳
⓴
㉑

2.1 Front panel

① POWER switch (STANDBY/ON)

The power switch is used to switch the unit between standby mode (the time from the internal clock is displayed) and on (full operational mode).

When the unit is turned on, the display shows Welcome!.

② Disc tray

Discs should be placed label side upwards in the disc tray.

When a disc is loaded into the unit, the display briefly shows TOC Reading., with a flashing period following the word “Reading” as the unit determines the contents of the disc.

③ OPEN/CLOSE key

Use this key to open and close the disc tray. The display shows OPEN when the tray is opening and CLOSE when the tray is closing.

④ RHSL key

Use this key when recording to set and adjust the start point of a recording session or the start of a track on the recorded disc. See 4.4, “Rehearsal” for details.

⑤ DIGITAL DIRECT

When this is active, any digital audio signals input to the unit bypass the frequency convertor and the digital volume control. See 4.3.4, “Digital direct” for details

⑥ INPUT SELECT

Repeated presses of this key cycle through the different audio inputs and the input combinations. See 4.1, “Input selection” for full details.

⑦ REMOTE SENSOR

This is not a control, but the sensor for the remote control unit. When using the remote control unit, make sure that there is a clear path from the remote control unit to the sensor.

⑧ Display

The display is used to show the current status of the unit, as well as the current menu settings. See 2.2, “Rear panel” for details.

⑨ MENU key

Repeated presses of this key cycle through the different menu items used to set parameters.

⑩ DISPLAY key

Repeated presses of this key cycle through the different time display modes. These modes differ, according to whether the unit is playing back or recording a disc.

⑪ MULTI JOG control

Use this control to select tracks for playback, to set parameter values, etc.

Typically, turning the control clockwise will increase a value, and turning it counterclockwise will decrease the value. There is no “end-stop” to the control—it may be turned continuously in either direction.

In addition, pushing the control acts like an “enter” key, i.e. the currently-displayed value or setting is confirmed.

⑫ TIMER switch (PLAY, OFF, REC)

Use this switch with the timer functions to control the behavior of the unit when the timer “on” time is reached. See 2.5, “Timer operations” for details.

⑬ PHONES jack and LEVEL control

Use a standard pair of stereo headphones, equipped with a 1/4-inch plug, with this jack.

Adjust the volume from the jack with the control (turn clockwise for higher volume).

⑭ REC LEVEL controls

Use these concentric controls to adjust the level of the analog audio signals received at the **ANALOG IN** jacks ㉓. Typically, turning one of these controls will cause the other to turn, but they can be turned independently of each other.

Use the lower (outer) control to adjust the left signal, and the upper (inner) control to adjust the right signal.

Turning the controls clockwise past the “**5**” position will boost the signals relative to their input level, and turning them counterclockwise to a position below “**5**” will cut the signals relative to their original input level.

⑮ FINALIZE key

Use this key to finalize recordable discs. See 1.2, “Finalizing” and 5.1, “Finalizing” for details.

⑯ ERASE key

Use this key (with CD-RW discs only) to erase tracks, or a whole disc, or to refresh a “bad” disc, or to unfinalize a finalized disc. See 5.2, “Erasing” for details.

⑰ STOP key

Use this key to stop playback or recording.

⑱ PLAY key

Use this key to start or resume playback or recording.

⑲ PAUSE key

Use this key to pause playback or recording.

⑳ RECORD key

Use this key to enter record ready mode (see 4, “Recording” for details) and also to enter manual track divisions (see 4.2.3, “Manual track division”).

㉑ SYNC REC key

Use this key to turn synchronized recording on and off (see 4.3.1, “Synchronized recording” for details).

㉒ FADER key

Use this key to start fade-in recordings or to start a fade-out when recording (see 4.3.6, “Fade-in and fade-out” for details).

2.2 Rear panel

㉓ ANALOG IN (L, R) (UNBALANCED)

These RCA jacks accept analog audio signals from suitably-equipped units (–10 dBV, unbalanced).

㉔ ANALOG OUT (L, R) (UNBALANCED)

These RCA jacks output analog audio signals (at –10 dBV levels, unbalanced) to suitably-equipped units.

㉕ DIGITAL COAXIAL (IN, OUT)

These two RCA jacks accept and output digital audio in standard consumer format (SPDIF).

㉖ DIGITAL OPTICAL (IN, OUT)

These two TOSLINK optical connectors accept and output digital audio using optical fiber connectors.

2.3 Remote control unit

❶ POWER key

Use this to turn the unit between off and standby modes (equivalent to ①).

❷ OPEN/CLOSE key

Opens and closes the disc tray (equivalent to ③).

❸ Number keys (0 through 10 and +10)

Use these keys for direct entry of track numbers when playing back, or making a programmed play list.

❹ CLEAR key

Use this key to clear a mistaken entry made with the number keys.

❺ REPEAT key

Use this key to change between the various repeat modes when playing back (see 3.3, “Repeat play”).

❻ DISPLAY key

Use this key to switch between the different display modes available (equivalent to ⑩).

❼ MENU key

Repeated presses of this key cycle through the different menu items used to set parameters (equivalent to ⑨).

❽ ENTER key

Pressing this key is equivalent to pressing the **MULTI JOG** control ⑪ to confirm an entry.

❾ FADER key

Use this key to start fade-in recordings or to start a fade-out when recording (equivalent to ㉒).

❿ MONITOR key

Use this key to output the signals received at the currently selected input(s) through the outputs when not recording or in record ready mode.

⓫ PLAY MODE key

Use this key to choose between the different playback modes (single play, continue play, etc.).

⓬ FINALIZE key

Use this key to finalize recordable discs. See 1.2, “Finalizing” and 5.1, “Finalizing” for details (equivalent to ⑮).

⓭ ERASE key

Use this key (with CD-RW discs only) to erase tracks, or a whole disc, or to refresh a “bad” disc, or to unfinalize a finalized disc. See 5.2, “Erasing” for details (equivalent to ⑯).

⓮ RECORD key

Use this key to enter record ready mode (equivalent to ⑳).

⓯ SYNC REC key

Use this key to turn synchronized recording on and off (equivalent to ㉑). See 4.3.1, “Synchronized recording” for details).

⓰ REC MUTE key

Use this key when recording, to mute the input signal for about four seconds and then to enter record ready mode (see 4.3.5, “Rec mute function”).

⓱ SEARCH keys

Use these keys to move the playback position inside a track.

⓲ SKIP keys

Use these keys to move between tracks when playing back (equivalent to turning ⑪). They are also used to set the parameter values, etc.

⓳ STOP key

Use this key to stop playback or recording (equivalent to ⑰).

⓴ PAUSE key

Use this key to pause playback or recording (equivalent to ⑲).

㉑ PLAY key

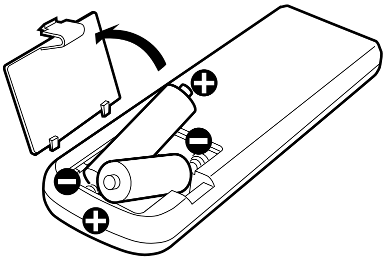
Use this key to start playback, or enter record mode from record ready mode (equivalent to ⑱).

2.3.1 Use of the remote control unit

When using the remote control unit, make sure:

- There is a clear unobstructed path between the remote control unit and the remote sensor ⑦ on the main unit.
- The remote control unit is located within 5 m (15 ft.) of the main unit, and is pointing at approximately right angles (±30°) to the front panel.

When fitting or replacing the batteries:



- Always make sure that both batteries are replaced together. Do not mix old and new batteries.
- Do not mix batteries of different types.
- Always make sure that the batteries are located with the correct polarity (the positive terminals of the batteries should match the **+** markings inside the battery compartment, and the negative terminals should match the **–** markings).
- If you are not going to use the remote control unit for an extended period of time, remove the batteries. Old batteries can leak, casing damage to the remote control unit.
- Always dispose of old batteries in the way recommended by your local garbage disposal authorities.

2.4 Connections

The following should be noted when you connect the unit to other equipment.

It is possible to connect the unit’s **DIGITAL COAXIAL IN** ㉕ and **DIGITAL OPTICAL IN** ㉖ to other equipment at the same time. However, the unit can receive signals from only one digital input at a time.

Audio signals output from the unit are output from the **ANALOG OUT** ㉔ jacks, as well as from the **DIGITAL COAXIAL OUT** jack ㉕ and **DIGITAL OPTICAL OUT** connector ㉖ simultaneously.

If the **ANALOG IN** jacks ㉓ are connected, it is possible to select a “mix mode”, where the **ANALOG IN** signals are mixed with the digital audio signals at either the **COAXIAL** or **OPTICAL** input. The digital audio signal is attenuated (cut) by 12 dB when this mix takes place.

2.5 Timer operations

The unit can use its internal clock to start and stop playback or recording at pre-determined times.

2.5.1 Setting the clock

1 **When the unit must be in standby mode (power connected but not turned on), and if the time has not been set, the display shows `Clock Adjust` in dimmed characters.**

2 **Press the MENU key (⑨ or ❼). The display lights, and shows `CLOCK ADJ ?`.**

3 **Press the MULTI JOG control ⑪ or the ENTER key on the remote control unit ❽. The “hours” value will start flashing.**

4 **Turn the MULTI JOG control or use the remote control unit SKIP keys ⓲ to set the current hour.**

5 **Press the MULTI JOG control or the ENTER key to confirm the hour setting. The “minutes” value starts flashing.**

NOTE

The unit uses the 24 hour “military” clock, where 1 p.m. is represented by 13, etc.

6 **Turn the MULTI JOG control or use the remote control unit SKIP keys to set the current minutes.**

7 **Press the MULTI JOG control or the ENTER key to confirm the minute setting and return to the standby mode. The unit now shows the current time in standby mode and the colon : between the hours and the minutes flash once per second.**

NOTE

If the unit is disconnected from the power supply for an extended period of time, the clock must be reset (the Clock Adjust message appears when power is reapplied). For this reason, if connecting the unit through a power distributor, we suggest that you use an unswitched outlet on the distributor.

2.5.2 Setting the timer

Once the clock has been set following the procedure above, the timer can be set.

The procedure below sets the time at which the unit will turn on and off.

1 **Make sure the unit is turned on and a disc is inserted.**

2 **Press the MENU key (⑨ or ❼) until the display shows TIMER ?.**

3 **Press the MULTI JOG control ⑪ or the ENTER key on the remote control unit ❽.**

4 **Turn the MULTI JOG control or use the remote control unit SKIP keys ⓲ to set the timer hour.**

NOTE

The unit uses the 24 hour “military” clock, where 1 p.m. is represented by 13, etc.

5 **Press the MULTI JOG control or the ENTER key to confirm the hour setting. The “minutes” value starts flashing.**

6 **Turn the MULTI JOG control or use the remote control unit SKIP keys to set the timer minutes.**

7 **Press the MULTI JOG control or the ENTER key to confirm the minutes setting.**

8 **After step 7 above, the display shows OFF 0:00.**

9 **Repeat steps 4 through 7 above to set the hours and minutes value.**

If the TIMER switch ⑫ is set to REC or PLAY, the display shows a small “clock” icon, to show that the unit is in timer mode.

2.5.3 Using the timer

When the start time has been set, slide the **TIMER** switch ⑫ to either the **PLAY** or **REC** position, depending on whether playback or recording is to start at the specified time.

The display shows a small clock icon to show the unit is in timer mode, in standby mode as well as in power-on mode. When the switch is set to **REC**, and a recordable disc is loaded, the REC indicator shows beside the clock icon.

When the unit is in standby mode and timer playback is selected, the unit will turn on and start playing at the specified time.

If recording is selected, the unit will turn itself on one minute before recording is due to start. Recording will start at the specified time at the first free location on the disc.

If the off time is set to a time greater than the length of the disc, and playback is selected, the unit will start playing at the start of the disc, and continue to the end.

If timer recording is selected, and the off time is set to a time which is greater than the remaining time on the disc, the unit will continue recording until there is no space left on the disc.

If the timer is set, and the unit is switched on at the timer start time, the timer operation will have no effect.

If the unit has been disconnected from the power supply so that the clock time has been lost, and the timer switch is set, the clock icon will flash rapidly in standby mode as an indication that the current timer setting is invalid.

NOTE

If the unit has recognized the current disc as non-recordable, Timer Error! appears when the timer switch is set to **REC***. If the timer is set with the switch at* **OFF** *or* **PLAY** *and a non-recordable disc loaded, the unit is then placed in standby, and the switch then set to the* **REC** *position, this error message will not appear (of course, recording will not take place under these conditions).*

The CD-RW700 can be used to play back CDs (including CD-R and CD-RW discs that have been recorded and finalized on other types of recorder), as well as CD-R discs and CD-RW discs that have been recorded on the unit.

3.1 Simple playback operations

1. **Press the OPEN/CLOSE key (③ or ❷) to open the disc tray.**
2. **Discs should be inserted in the tray with the label uppermost.**
3. **Press the OPEN/CLOSE key (③ or ❷) to close the tray again.**

When a disc is loaded into the unit, the display briefly shows `TOC Reading.`, with a flashing period following the word "Reading" as the unit determines the contents of the disc. When the TOC has been read, the display shows (at the top left of the display) one of the following:

CD	A commercial pre-recorded CD or a finalized CD-R
CD-R, NO TOC	A CD-R which has yet to be finalized
CD-RW, NO TOC	A CD-RW which has yet to be finalized
CD-RW	A finalized CD-RW (this differs from a CD, as it may be unfinalized and erased)
[blank]	Unusable or unreadable disc or no disc

1. **Press the PLAY key (⑱ or ㉑) to start playback**
2. **Press the STOP key (⑰ or ⓳) to stop playback**
3. **Press the PAUSE key (⑲ or ⓴) to pause playback temporarily.**

The track number indicators at the bottom of the display light, to show the available unplayed tracks. If there are thirteen tracks on the disc, for example, indicators 1 through 13 will light. If there are more than 20 tracks on a disc, the track number OVER indicator (at the bottom of the display) will light.

3.1.1 Time display modes

When playing back a CD (or in pause mode), there are four different time display modes.

In each mode, the current track number is given, followed by the time, as given below, e.g.

`10Tr 0:48`

Press the **DISPLAY** key (⑩ or ❻) to cycle through these time display modes, as described below:

Display shows	Meaning
TOTAL	Total elapsed time of the disc
[blank]	Elapsed time of the current track
TOTAL REMAIN	Time remaining of the current disc
REMAIN	Time remaining of the current track

NOTE

If programmed playback has been selected (3.2, "Programmed order playback"), the elapsed time and total remaining time refer to the programmed material, not the whole disc.

3.1.2 Track search

Use the **MULTI JOG** control ⑪ or the **SKIP** keys on the remote control unit ⓲ to jump forward or backward by one track at a time, as shown on the display.

Note that going forward from the last track of the disc will "wrap round" to track 1, and going backward from the first track of the disc will "wrap round" to the last track of the disc.

The track number indicators at the bottom of the display shows the selected track (the lit indicator at the left of the row of indicators).

It is also possible to use the number keys on the remote control unit ❸ to jump directly to a particular track:

1. **To play tracks 1 through 9, simply press the appropriate key (1 through 9).**
2. **To play a track whose number is higher than 9, press the +10 key an appropriate number of times to set the "tens" digit of the track number, followed by a single key to set the "units" digit. For example:**

   Track 13 = +10, 3

   Track 30 = +10, +10, +10, 0

On the remote control unit only, the **SEARCH** *keys* ⓱ *can be used to "fast-forward" and "fast-rewind" through the disc.*

3.1.3 Playback modes

The unit can be set to play back a single track, the whole disc in the order in which it was created, the tracks of the disc in a random (shuffled) order, or a programmed order (see 3.2, "Programmed order playback").

NOTE

Selecting the playback mode is only possible using the remote control unit. It is not possible to select the playback mode using the main unit alone.

1 Press the PLAY MODE key ⓫ repeatedly to cycle the display through the following:

- Program (see 3.2, "Programmed order playback" below). The red PROGRAM display indicator lights in this mode.
- Shuffle (random track order). In this mode, each track is played once before the disc repeats. Using the track search functions will move backwards or forwards through the random order, e.g. if track 7 is followed by track 10, it is possible to use the track search keys to go back to track 7, which is remembered as being before immediately before track 10. The red SHUFFLE display indicator lights in this mode.
- Continue. Normal playback mode. Programming is cancelled, and playback continues in the normal order.

The selected mode is shown in large characters disappears from the display a few seconds after selecting the mode, and the display returns to its previous mode.

3.2 Programmed order playback

The steps to set up to 25 program steps (tracks) to be played back in a specified order are as follows:

1 Press the PLAY MODE key ⓫ until the display shows Program.

2 The display changes to show
0Tr 0:00 00,
meaning that no program steps have yet had tracks assigned to them.

3 Use the number keys to enter a track number (including the +10 key to enter values greater than 9, as described in 3.1.2, "Track search"). There is no need to press ENTER.

The track entered is assigned to the program step, and the display changes, showing the track number, the total time of the program so far, and the step number, e.g. 2Tr14:56 03.

4 The next program step can then be entered.

NOTE

Steps 3 and 4 above can be carried out from the main unit by turning and pushing the **MULTI JOG** *control ⑪.*

5 Repeat steps 3 and 4 above until the program is complete.

6 Press the PLAY key (⑱ or ㉑) to start playback at the first program step.

The display shows the track, the current time display, and the program step number.

While in programmed playback mode, using the SKIP keys ⓲ or the MULTI JOG control ⑪ moves backwards and forwards in the order set in the program.

Use the **PAUSE** key (⑲ or ⓴) to halt playback temporarily or the **STOP** key (⑰ or ⓳) to stop playback of the program.

If the **STOP** key is used, when playback is restarted, it starts from the first program step.

Clear the program by pressing **PLAY MODE** ⓫ until the display shows Continue. The program is also cleared when the disc tray is opened, and if the **STOP** key is pressed when playback is stopped in programmed order mode.

The programmed play order is not memorized when power is turned off.

If an attempt is made to enter more than 25 program steps, the message PGM Full ! appears on the display.

3.3 Repeat play

The unit can repeat the whole of the disc (or program) or the current track only.

NOTE

Selecting the repeat mode is only possible using the remote control unit. It is not possible to select the repeat mode using the main unit alone.

1 The unit should be in playback, pause or stop mode.

2 Continue to press the REPEAT key ❺ so that the display shows Repeat 1 (the REPEAT 1 indicator lights) to repeat one track, or Repeat ALL (the REPEAT ALL indicator lights) to repeat the whole disc or program.

3 If playback has not started, restart it.

To cancel repeat mode, press the **REPEAT** key so that the display shows Repeat OFF and the REPEAT indicator goes out.

Before you start recording using the CD-RW700, make sure that you understand the following key points:

- Once you have recorded on a CD-R disc, the data cannot be erased from it.
- You can add tracks to an unfinalized recordable disc. Once finalized, a CD-R disc is “fixed” and behaves in the same way as a pre-recorded disc. Finalized CD-RW discs, however, can be erased, refreshed and re-recorded.

4.1 Input selection

To select the input to be recorded, press the **INPUT SELECT** key on the main unit ⑥.

Repeated presses of the key cycle through the following options:

`Analog In`	Analog in
`Optical In`	Optical digital in
`Coaxial In`	Coaxial digital in
`Opt +Analog`	Optical digital and analog, mixed
`Coax +Analog`	Coaxial digital and analog, mixed

In the case of either the coaxial or optical inputs being selected, either alone, or together with the analog inputs, the boxed DIGITAL indicator shows on the display, and the appropriate indicator: COAXIAL or OPTICAL, lights on the display.

If the analog signal is selected, either alone, or together with either of the digital inputs, the boxed ANALOG indicator lights on the display.

4.1.1 Signal volume

The signals from either of the digital inputs (coaxial or optical) can be mixed with the signals from the analog inputs. The signal path is as follows (this description applies, no matter how many inputs are being mixed):

- The analog signals received at the **ANALOG IN** jacks are controlled with the **REC LEVEL** controls ⑭.
- Any input analog signals are then mixed with the digital signals received at either the **COAXIAL** or **OPTICAL IN** (㉕ or ㉖). If analog and digital signals are being mixed, the digital signals are attenuated by 12 dB. The level of the analog/digital mix is then controlled using the digital volume menu, as described below.

1 **Press the RECORD key (⑳ or ⓮) to place the unit in record ready mode.**

2 **Repeatedly press the MENU key (⑨ or ❼) until the display shows `VOLUME XXXdB` where `XXX` is the current setting.**

3 **Use the MULTI JOG control ⑪ or the SKIP keys ⓲ to adjust the input volume between `-60dB` and `+18db` (relative to the original input level). There is also a `----` setting which represents a complete signal cut.**

NOTE

If digital direct is selected, as described below (4.3.4, “Digital direct”), the digital volume is bypassed and the menu item does not appear.

4.1.2 Input monitoring

When the unit is in record ready, or record mode, the input signal is output from the **OUT** jacks (digital and analog), allowing you to hear the input signal.

The meters also show the current signal level.

However, it is only possible to record, or to enter record ready mode, when a recordable disc is inserted. To allow monitoring of the input signal at other times, or when the unit is stopped, use the **MONITOR** key on the remote control unit ❿.

The **RECORD** key ⑳ on the main unit will also enter monitor mode from stop mode when there is no recordable disc in the unit (when there is a recordable disc in the unit, pressing the **RECORD** key enters record ready mode).

The word `MONITOR` appears on the display to show that the output signal is an echo of the input signal(s). The meters show the current input level.

Exit the monitor mode by using the **STOP** key.

4.1.3 Frequency conversion

The sampling frequency for CDs is 44.1 kHz. The CD-RW700 will always record (and play back) CDs at this frequency.

However, if recording through a digital connection (either coaxial or optical) and the source has been recorded at a frequency other than 44.1 kHz, or is being played back at a non-standard speed using a varispeed unit (more than 1% away from the standard), the CD-RW700’s internal frequency converter will automatically convert the incoming frequency to 44.1 kHz.

NOTE

If digital direct is selected, as described below (4.3.4, “Digital direct”), the frequency convertor is bypassed. In this case, recording from digitally-connected devices is only possible when the source frequency is 44.1 kHz. If an attempt is made to record from a digitally-connected source at a different frequency, the error message Not Fs44.1k! *is displayed.*

4.2 Basic recording

NOTE

According to the “Red Book” (the specification for audio CDs), a track cannot be less than four seconds in length, and there can be a maximum of 99 tracks on an audio CD. Bear these limitations in mind when recording.

If you press either the **STOP** *or the* **PAUSE** *key within four seconds of starting recording, the unit will continue recording until four seconds have elapsed (since the start of recording) and then stop or pause as appropriate.*

After selecting the input source(s) (4.1, “Input selection”), it is possible to start recording.

The basic record procedure is as follows:

1 **Load a recordable disc is into the unit. As the disc is loaded, the display shows** TOC Reading.

When the disc has been loaded, the display indicators show CD-R, NO TOC, **or** CD-RW, NO TOC, **as appropriate.**

2 **Press the RECORD key (⑳ or ⓮). The unit enters record ready mode, as shown by the** REC **and pause indicators on the display. As the unit enters record ready mode, the display shows** Now OPC **(Optimum Power Control), as the unit prepares to record on the disc.**

3 **Adjust the level of the source(s), following the guidelines in 4.1.1, “Signal volume”.**

NOTE

The red OVER *indicators on the meters should never light. Unlike analog equipment, digital audio units produce extremely unpleasant sounds when distorted, and there is no “headroom” after the* 0 *mark. If recording digitally from a commercially-produced CD, the volume set in the menu should be* 0dB*. This will maintain all the dynamic range of the source CD without clipping. Only boost the signal if the source is exceptionally quiet.*

An analog signal input at nominal level, with no cut or boost from the volume controls, is equivalent to a reading of –16 dB on the meters (on other words, 0 is equivalent to 16 dBFS).

4 **Press the PLAY key (⑱ or ㉑) or the PAUSE key (⑲ or ⓳) to start recording.**

Note that there is no need to “cue up” the recording position—the unit always finds the next available location on the disc and start recording to it.

5 **Press the PAUSE key (⑲ or ⓳) to pause recording temporarily. The pause indicator on the display shows this.**

Pressing PAUSE or PLAY again restarts recording with a new track number.

6 **Press STOP (⑰ or ⓳) to stop recording. When recording stops, the display shows** PMA Writing **(Program Memory Area) and the** REC **indicator flashes for a few seconds as the unit writes to the disc.**

NOTE

While the “PMA Writing” message is displayed, all keys are disabled, and the unit is actually writing to disc. Do not disconnect the power or subject the unit to severe vibration or shocks at this time, as this will prevent proper recording of the information.

When this message disappears, you must press the RECORD key (step 2) before restarting the recording. Recording restarts with a new track number.

NOTE

Remember that every time recording is paused or stopped, when recording is restarted, a new track will always be started. It is not possible to record in two “stages” within one track.

4.2.1 Playing back tracks

Although an unfinalized disc cannot be played back on an ordinary CD player, the CD-RW700 can play back tracks that have been recorded.

Use the **MULTI JOG** control ⑪ or the **SKIP** keys ⓲ to select a track for playback.

4.2.2 Time display while recording

While a recordable disc is being played back, the four time display modes available in normal playback as described in 3.1.1, “Time display modes”: TOTAL, TOTAL REMAIN, track REMAIN and elapsed track time.

However, during recording there are fewer options available (obviously, the unit cannot see into the future and know how much time remains of the track which is being currently recorded!).

When in record or record ready mode, repeated presses of the **DISPLAY** key (⑩ or ❻) cycle between displaying the remaining time left for recording on the disc (TOTAL REMAIN) and the elapsed time of the track currently being recorded (no indicator lit).

When playback is stopped at the start of a recordable disc, there are only two options available: TOTAL (the total time recorded so far on the disc) and TOTAL REMAIN (the total time available on the disc for further recording).

4.2.3 Manual track division

It is possible to divide the recording into tracks “on the fly” while recording.

While recording is taking place, press the **RECORD** key (⑳ or ⓮) . The current track number is incremented by one.

For details of automatic track division while recording, see 4.3.2, “Automatic track division”.

NOTE

According to the “Red Book” (the specification for audio CDs), a track cannot be less than four seconds in length, and there can be a maximum of 99 tracks on an audio CD. Bear these limitations in mind when adding track divisions.

4.2.4 Restricting the number of tracks

When making a recording using the CD-RW700, it is possible to restrict the number of tracks that may be recorded at one time. This applies whether the tracks are being divided manually, as described above (4.2.3, “Manual track division”) or being divided automatically (4.3.2, “Automatic track division”).

At any time within record or record ready mode, turn the **MULTI JOG** control ⑪ or use the **SKIP** keys ⓲.

This sets the number of remaining tracks, as shown to the right of the time display. The maximum number here is limited by the standard limit of 99 tracks, and the number of tracks currently recorded. In other words, if 8 tracks have already been recorded on the disc, the maximum value that can be set here is 91 (91+8=99). If 49 tracks have already been recorded on the disc, the maximum value that can be set here is 50 (49+50=99).

Every time a track division is made, the number shown at the right of the display goes down by one. When the number shown is 1 , and a track division is made, the display shows PMA Writing and the unit enters stop mode.

The number of remaining tracks can be changed during recording using the **MULTI JOG** control.

4.2.5 Full disc

If recording a long program, and the number of tracks has not been restricted, when there is no space remaining on the disc for recording, as shown by the time display (see 4.2.2, “Time display while recording”), just before the end of the disc is reached, the unit starts a fade-out over a few seconds (the display shows FADE OUT).

When the fade-out has finished, the display shows Disc Full!! for a few seconds, and the unit then goes into stop mode (the display shows PMA Writing).

If the disc is a CD-RW, the final track can be erased, if required (see 5.2.1, “Erasing tracks”) and the disc then finalized. If the disc is a CD-R, it must be finalized prior to use (see 5.1, “Finalizing”).

4.3 Advanced recording

These sections cover more advanced recording techniques.

4.3.1 Synchronized recording

Synchronized recording means that recording will begin automatically when a signal is received, and will stop when the signal ends.

To enable or disable synchronized recording:

1 **A recordable disc must be loaded.**

2 **With the unit in record ready or record mode, press the SYNC REC key (㉑or ⓯).**

The display shows SYNC ON briefly, and then changes to the track and time display. When synchronized recording is enabled, the SYNC indicator lights in the display.

The SYNC REC key is also used to turn off synchronized recording (the display briefly shows SYNC OFF and then returns to the track and time display).

The signal level which may be used with synchronized recording operations (here called the “trigger level” or “threshold”) is set as below:

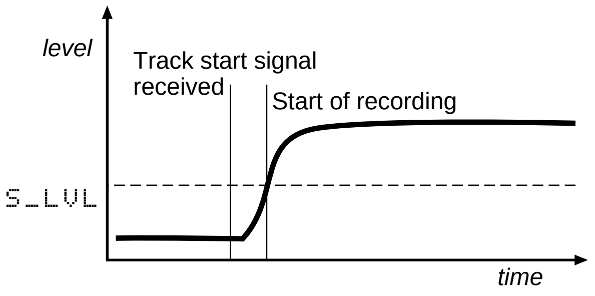
1 **With a recordable disc loaded, press the MENU key (⑨ or ❼) until the display shows: `S_LVL>> xxdB`, where `xx` is the current value.**

2 **Turn the MULTI JOG control ⑪ or use the SKIP keys ⓲ to change the value between `–24`, `–30`, `–36`, `–42`, `–48`, `–54`, `–60`, `–66`, and `–72` dB.**

Low numerical values of this setting (towards –24) mean that it takes a louder signal to trigger the start of the synchronized recording. A high numerical value (towards –72) means that a relatively quiet signal will trigger the start of recording.

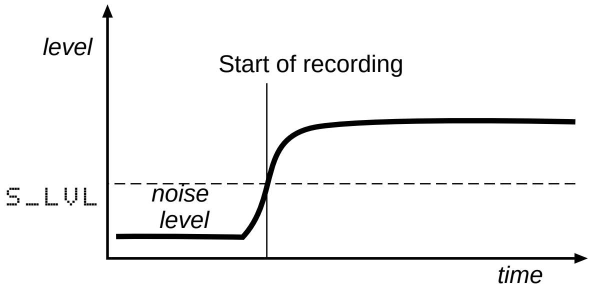
The conditions under which an input signal is used as the trigger to start recording are:

- when recording digitally, a digital start signal is detected, and audio is subsequently detected



or

- the sound level rises above the threshold (see below) after having been below the threshold for one second or more. Note that the diagram below represents a special case (the source is stopped or paused) of the more general condition, as explained later:



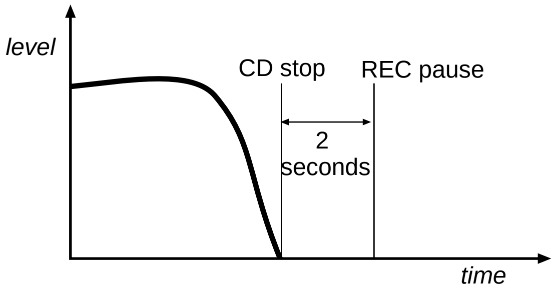
To start synchronized recording when the source is stopped or in pause mode:

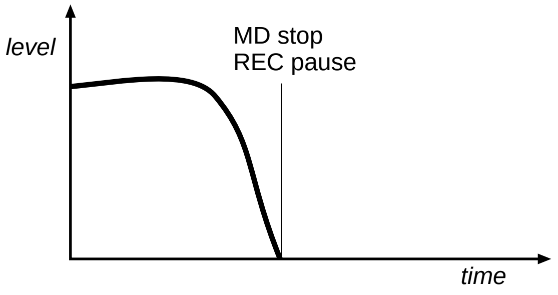
1 **The unit must be in record ready mode, and the SYNC indicator in the display must be lit, as explained above.**

2 **Start playing the source. The pause indicator disappears from the display, and the unit starts recording.**

The conditions under which synchronized recording stops are:

- when recording digitally, an appropriate digital signal is detected. If the signal is detected from a CD, the unit enters record ready mode after 2 seconds. If the signal is detected from an MD, the unit enters record ready mode immediately:

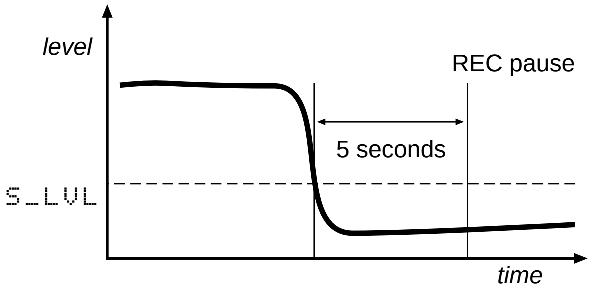




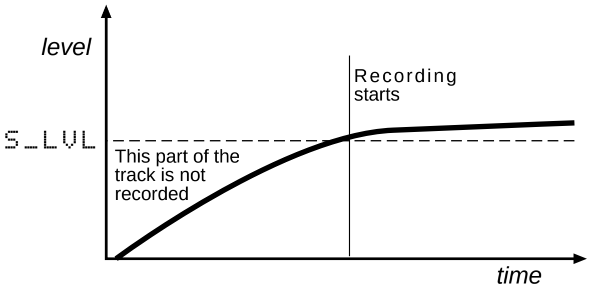
Note that the trigger level is not used here.

or

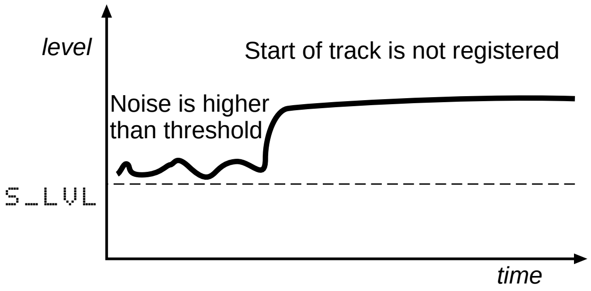
- the sound level drops below the threshold level for five seconds



If you set the threshold value so that only high-level signals trigger recording, remember that this will not record the start of a piece with a slow fade-in:



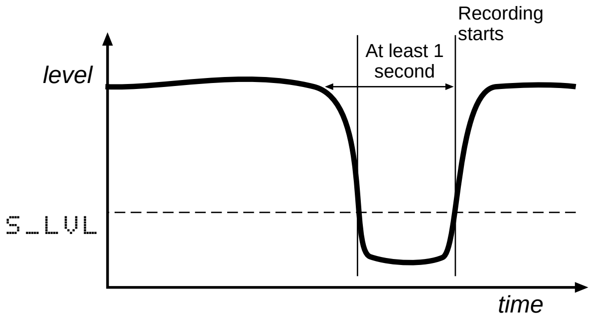
Also remember that when recording from an analog source, the threshold should be set so that it is higher than the “noise floor” (the base level of hiss, pops and crackles which may come from an older vinyl recording, for example). If the "noise floor" is higher than the threshold, synchronized recording will not take place (the rehearsal facility can be used here (4.4, "Rehearsal") to help):



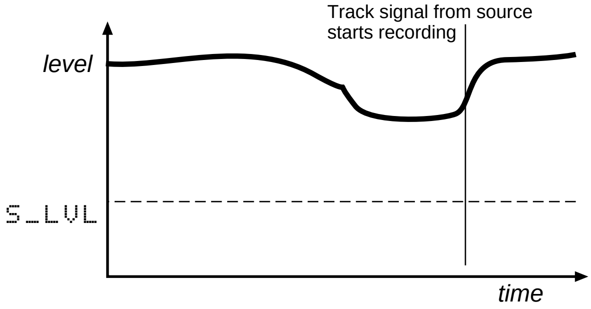
Because synchronized recording can be turned on and off while recording is taking place, it is therefore possible to record with a synchronized automatic start, and a manual end, or *vice versa*.

When the source material is being played back before synchronized recording is enabled, the conditions under which synchronized recording will start (after synchronized recording has been enabled) are:

When the signal drops below the threshold for one second or more and then rises above the threshold again (this is the general condition, of which starting synchronized recording from a stopped or paused source is a special case):



Alternatively, when recording digitally from sources (MD, CD and DAT) which output track divisions, synchronized recording starts when a track division is received, regardless of the level:



To stop synchronized recording:

1. **The unit must be in record mode, and the** SYNC **indicator in the display must be lit.**
2. **Stop the source. After a few seconds, the unit enters record ready mode. The time between the stopping of the source and entering record ready mode is used to record a gap at the end of the track.**
3. **If no changes are made to the unit's controls, restarting the source restarts recording, as the unit is still in sync mode and will start recording again when the source is detected.**

NOTE

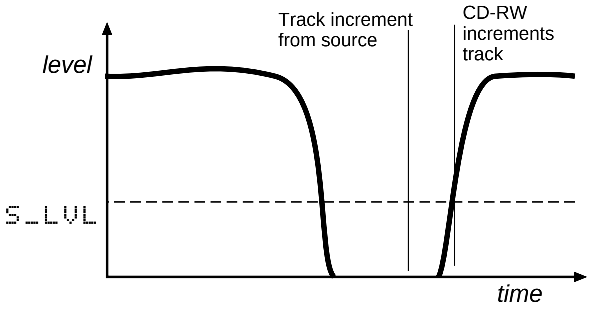
When recording in sync mode, you may notice a slight delay between starting the source, and the CD-RW700 responding. This is due to the way in which the unit works, storing the data in a memory buffer before writing it to the disc. The resulting delay is therefore not a cause for concern—all audio data received between the start and end points will be recorded.

4.3.2 Automatic track division

The trigger level, as set above (4.3.1, "Synchronized recording" can also be used to divide tracks if the A-TRACK (auto track) function is enabled.

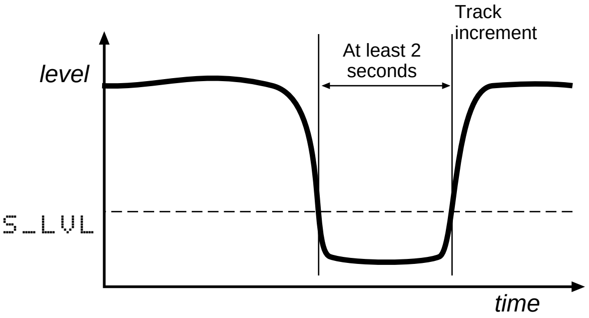
When this function is enabled, the unit automatically inserts a track division in the recorded material when:

- recording digitally from MD, CD or DAT, an increment signal is detected and the sound level rises above the threshold:



or

- increment signals are unavailable (e.g. analog recording or digital sources other than MD. CD or DAT) and the sound level has dropped below the threshold for more than two seconds, and sound has re-started



To enable or disable automatic track division:

1. **With a recordable disc inserted, and with the unit in record ready mode, press the MENU key (⑨ or ❼) repeatedly until A_TRACK>XX is shown, where XX may be ON or OFF.**
2. **Turn the MULTI JOG control ⑪ or use the SKIP keys ⓲ to change the value to ON or OFF.**

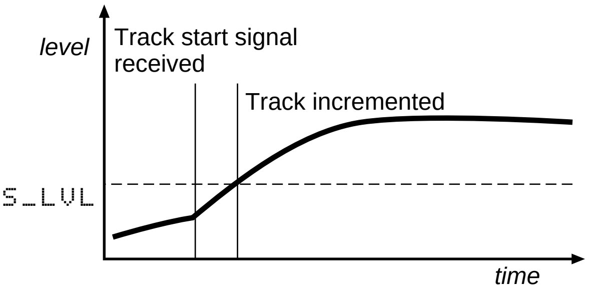
   The A-TRACK indicator at the top right of the display lights when the auto track function is enabled.

Automatic track division can be turned on or off while recording is taking place.

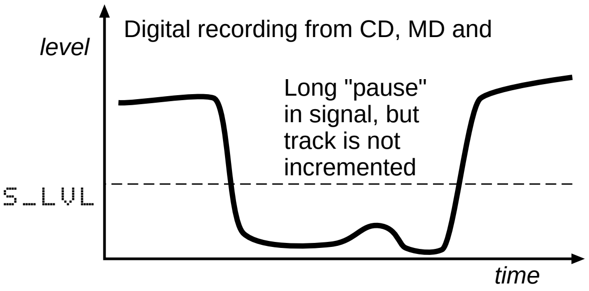
NOTE

When recording certain types of material (e.g. certain types of classical music or spoken word) through the analog connections, since the track divisions are determined by relative silence, it is possible that spurious track divisions will be inserted. Under these conditions, it may be better to use manual track division (4.2.3, “Manual track division”).

However, if recording through a digital connection, the track number will be incremented only when a track division signal is received, followed by a rise in volume.



A fall and subsequent rise in volume alone when recording digitally from CD, MD or DAT is not sufficient to trigger an automatic track increment.



You may find it helpful to use the track restriction facility (see 4.2.4, “Restricting the number of tracks”) together with the automatic track division.

For instance, if you have recorded two albums on one DAT tape, and you wish to make two separate CDs from this tape, you can use the **MULTI JOG** control to set the number of tracks to the number of tracks comprising the first album.

When the final track of the first album has been recorded, the unit automatically performs the PMA writing procedure and enters stop mode.

4.3.4 Digital direct

As mentioned earlier, the digital direct mode bypasses some of the input circuitry of the CD-RW700 (the frequency convertor and the digital level control) to write directly to the disc.

To enable or disable digital direct recording:

1. **Ensure that the input source is set to DIGITAL (either COAXIAL or OPTICAL).**
2. **Press the DIGITAL DIRECT key ⑤ (main unit only). The display shows D_Direct XX where XX is ON or OFF.**

3. **Press the DIGITAL DIRECT key to toggle between ON and OFF.**

   When digital direct recording is enabled, the DIGITAL indicator in the display flashes slowly.

NOTE

As mentioned earlier, the digital direct feature bypasses the frequency convertor. Recording is therefore only possible from sources recorded with 44.1 kHz sampling frequency.

4.3.5 Rec mute function

The rec mute function allows instant cut-off of the input signal, and records four seconds of silence on the disc.

1. **While recording is in progress, press the REC MUTE key ⓰.**

   The display shows REC MUTE, the meters stop displaying the current level and the monitored sound is cut.

2. **After about four seconds, the unit enters record ready mode.**

NOTE

This function is only available from the remote control unit. It is not possible to use this function with the main unit only.

4.3.6 Fade-in and fade-out

To provide smooth beginnings and endings to recorded tracks, the unit provides fade facilities.

Fade-ins are from –∞ (silence) to the set input level (4.1.1, “Signal volume”) and fade-outs go from the current input volume to silence.

The fade-in and fade-out times can be set independently in increments of one second, from 1 second to 24 seconds.

To set the fade times:

1. **Press the MENU key (⑨ or ❼) repeatedly until the display shows FADE IN > XXS, where XX is the current fade-in time, in seconds.**

2. **Turn the MULTI JOG control ⑪ or use the SKIP keys ⓲to change the value of the fade-in time.**

3. **Press the MENU key again so that the display shows FADE OUT> XXS, where XX is the current fade-out time, in seconds.**

4. **Turn the MULTI JOG control or use the SKIP keys to change the value of the fade-out time.**

   It is possible to set the fade-in time only (perform step 5 after step 2) or the fade-out time only (in step 1, press the MENU key until the fade-out time menu item is shown).

To perform the fade-in:

1. **With a recordable disc loaded, press the RECORD key (⑳ or ⓮) so that the unit enters record ready mode.**

2. **Start the source material, and press the FADER key (㉒ or ❾).**

   The display shows FADE IN for the time specified in the menu item above.

   As the fade progresses, the monitored signal gradually becomes louder, reflecting the level recorded on disc.

To perform a fade-out:

1. **While recording, press the FADER key (㉒ or ❾).**

   The display shows FADE OUT for the time specified in the menu above.

   As the fade progresses, the monitored signal gradually becomes quieter, reflecting the level recorded on disc.

2. **When the end of the fade-out time is reached, the unit goes into record ready mode. Recording can be restarted with the FADER key or the PLAY or PAUSE keys.**

The fade-in and fade-out times can be reset to new values while recording is in progress.

Fades are also possible in digital direct mode (4.3.4, “Digital direct”).

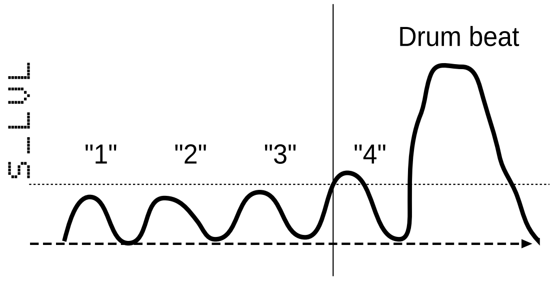
4.4 Rehearsal

There are two areas in which the unit’s rehearsal facilities can be used: synchronized recording (4.3.1, “Synchronized recording”) and automatic track division (4.3.2, “Automatic track division”).

The purpose of the rehearsal facilities is to allow you to override the automatic synchronized recording and track division facilities where necessary.

Here is an example where this facility may be of value:

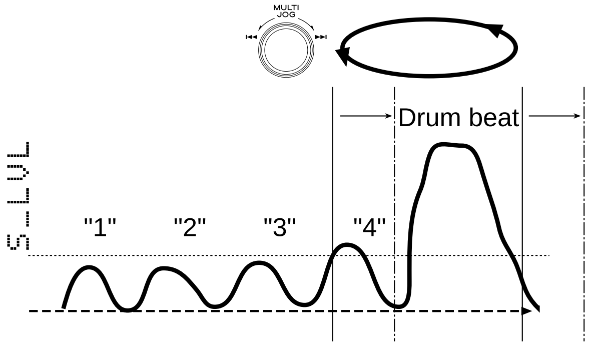
A piece has been recorded on DAT, with a verbal count-in immediately prior to the first drum beat. This count-in is not to be recorded as a part of the CD track, but the threshold level has been set to a level where the fourth count will trigger recording (–24 dB — this is the limit of the threshold sensitivity and cannot be adjusted further).



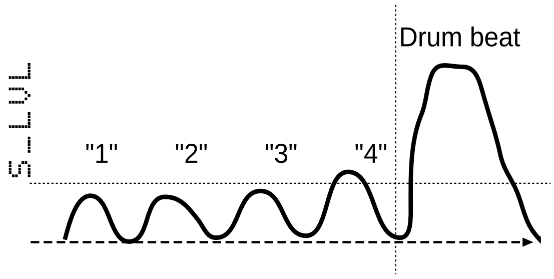
Sync recording starts here

When the rehearsal facility is used, the source is cued up and played, and the first few seconds of the track are sampled and looped. The position of the loop can be adjusted.

This section is looped. The **MULTI JOG** control adjusts the position of the loop



When this offset has been set, synchronized recording takes place with the start occurring at the new location:



Sync recording starts at new point

This example shows how the start time can be moved forward, in order to eliminate unwanted “false starts”. It is also possible to “slip” the start time backwards, in order to record a quiet passage that would otherwise not trigger the sync recording facility (this is done by the use of a memory buffer).

As well as being used for changing the start time of synchronized recording, the start of an automatic track division can also be trimmed backwards and forwards in a similar way.

Both the sync start trim and the track start trim can take the same value and can be used in the same recording.

The maximum trim value is 125 frames on either side of the original point (represented by 0), where there are 75 frames in a second (i.e. 1.67 seconds).

4.4.1 Rehearsal for synchronized recording

To perform a synchronized recording, using the rehearsal mode to slip the start of the track forwards or backwards:

1. **Load a recordable disc into the unit and make sure the unit is in stop mode.**
2. **Cue the source material to the start of the source track to be recorded (i.e. in the example above, this would be just before the count-in).**
3. **Press the RHSL key ④ so that the display shows Waiting Sync.**
4. **Start playing the source. When the start of the track is detected (as explained in 4.3.1, “Synchronized recording”), the display changes to Sync RHSL.**
5. **If the rehearsal does not start at the region of the true start point, press the RHSL key once again to “re-arm” the trigger.**
6. **After nine seconds, the message changes from Sync RHSL, and the sampled data loops in a four-second loop. The display shows Sync 0:0X, where X is the current second of the loop.**
7. **Stop the source.**
8. **Turn the MULTI JOG control ⑪ to adjust the point of the loop. The display shows: TRIM = +XXXf or TRIM = -XXXf, where XXX can be a number between 0 and 125 and changes back to the Sync 0:0x display when the loop restarts.**

   Every time the trim value is changed, the loop starts replaying from the new start point.
9. **When the new start point has been set, press the STOP key (⑰ or ⓳).**
10. **Cue up the source material to the original point set in step 2.**
11. **Press the RECORD key (⑳ or ⓮) to enter record ready mode, and then press the SYNC REC key (㉑ or ⓯).**

12 Start playing the source material. The unit starts recording, with the start point being determined by the trim value.

NOTE

If a trim setting is made for the sync start, this will apply to all sync starts and track divisions made in the future until the value is reset using the procedure above.

4.4.2 Rehearsal for track division

In the same way that the automatic threshold may produce a false trigger, starting recording at the wrong point, the automatic threshold may produce false track divisions.

By rehearsing the point at which the track division is to be made, it is possible to trim the division point so that it corresponds to a the track division which is actually required.

1 Load a recordable disc into the unit and make sure the unit is in stop mode.

2 If the unit is not already in auto track mode, set it so that the A-TRACK indicator is lit on the display (4.3.2, “Automatic track division”).

3 Cue the source material to a little before the point of the planned track division.

4 Press the RHSL key ④ twice so that the display shows Waiting Inc.

If the auto track mode has not previously been selected, the display shows No Inc Mode when the key is pressed.

5 Start playing the source. When the start of the track is detected (as explained in 4.3.1, “Synchronized recording”), the display changes to Inc RHSL.

6 If the rehearsal does not start at the region of the true track division point, press the RHSL key once again to “re-arm” the trigger.

7 After nine seconds, the message changes from Inc RHSL, and the sampled data loops in a four-second loop. The display shows Inc 0:0X, where X is the current second of the loop.

8 Stop the source.

9 Turn the MULTI JOG control ⑪ to adjust the point of the loop. The display shows:
TRIM = +XXXf or
TRIM = -XXXf, where XXX can be a number between 0 and 125 and changes back to the Inc 0:0x display when the loop restarts.

10 When the track division point has been fixed, press the STOP key (⑰ or ⓳).

11 Press the RECORD key (⑳ or ⓮) to enter record ready mode. The synchronized recording mode can be used here, and so the SYNC REC key (㉑ or ⓯) may be pressed to prepare for synchronized recording.

12 Cue up the source material to the point where recording is to start.

13 Start playing the source material. If sync recording is not being used, start recording on the CD-RW700 (otherwise recording starts automatically).

When the track division point (plus or minus the trim value set earlier) is reached, the recording will divide into tracks.

Note that if a positive trim value is set (i.e. the start point of the track is “slipped” to a position past the “normal” start), the counter shows a minus value (e.g. -0.02) for the period between the “normal” and the actual adjusted start position. This minus value is also displayed when recording starts, if sync recording has been enabled.

NOTE

If a trim setting is made for the track division, this will apply to all track divisions and sync starts made in the future until the value is reset using the procedure above.

5 - After recording

This section deals with the actions taken after recording: finalizing for CD-R and CD-RW discs, and erase operations for CD-RW discs.

Once finalized, CD-R discs are truly final—no more can be recorded on them. By contrast, CD-RW discs can be “unfinalized” using the CD-RW700, i.e. their TOC can be deleted, and if there is space, further material can be recorded on them. Even if the disc is full, tracks can be erased following the unfinalize process, and new material can be recorded.

5.1 Finalizing

As has been explained earlier (1.2, “Finalizing”), in order for a disc to become a standard CD, it must have a Table of Contents (TOC) written to it. This process is known as finalizing.

An unfinalized disc can always be distinguished on the CD-RW700 by the NO TOC indicator under the disc type on the display being lit when the disc is inserted.

To finalize a disc:

1 **With the unfinalized (recordable) disc loaded, and the unit in stop mode, press the FINALIZE key (⑮ or ⓬).**

   The display shows Finalize OK?.

2 **If you do not want to proceed with the finalizing process, press any other key (such as the STOP key).**

3 **To start the finalizing process, press the MULTI JOG control ⑪ or press the ENTER key ❽.**

4 **If finalization has been chosen, the display changes. The NO TOC indicator goes out, and the word** TOC **and the REC indicator flashes in the display. The unit starts counting down from a time shown on the display (about 1 minute 10 seconds).**

5 **After finalization, a CD-RW NO TOC changes to CD-RW and a CD-R NO TOC changes to a CD.**

NOTE

Although CD-R discs finalized on the CD-RW700 may be replayed on ordinary CD players, remember that finalized CD-RW discs may not play on CD players.

5.1.1 Unfinalizing CD-RW discs

If a CD-RW disc has been finalized, it is possible to erase the Table of Contents in the following way:

1 **With the finalized disc loaded and the unit in stop mode, press the ERASE key (⑯ or ⓭).**

2 **If the display does not show the word** Unfinalize?**, turn the MULTI JOG control ⑪ or use the SKIP keys ⓲ until it is displayed.**

3 **If you do not want to proceed with the unfinalizing process, press any other key (such as the STOP key).**

4 **To start the unfinalizing process, press the MULTI JOG control ⑪ or press the ENTER key ❽.**

5 **If unfinalization has been chosen, the NO TOC indicator starts flashing in the display, and the word** ERASE **also starts flashing. The unit starts counting down from a time shown on the display (depending on the disc and material, this value will be between 50 seconds and 1 minute 30 seconds).**

6 **After unfinalization, a CD-RW indicator changes to a CD-RW NO TOC indicator.**

5.2 Erasing

Unfinalizing (described above) is a special case of erasing a disc. It is also possible to erase a track or tracks, starting with the last-recorded track, a whole disc, or to “refresh” a disc.

NOTE

Erase procedures are possible only on a CD-RW disc. Even if a CD-RW disc has been finalized, selecting an erase procedure will automatically unfinalize it prior to erasure. The disc may then have to be re-finalized. Remember that it is not possible to perform any erase procedures on a CD-R disc.

5.2.1 Erasing tracks

It is possible to erase the last recorded track, the last two recorded tracks, the last three recorded tracks, etc. It is not, however, possible to erase tracks in the “middle” of the disc.

1 **With the disc loaded, and the unit in stop mode, press the ERASE key (⑯ or ⓭).**

2 **If the display does not show** ERASE XX-XX?**, where** XX **is the number of the last track recorded on the disc, turn the MULTI JOG control ⑪ or use the SKIP keys ⓲ until it is displayed.**

3 **Turn the MULTI JOG control or use the SKIP keys to change the number of the first track to be erased. The highest value that this number can take is the last track recorded (e.g. if 17 tracks have been recorded on the disc, 17-17 means that the last track only, 17, will be erased). The lowest value that this number can take is 2 (if this number was 1, the whole disc would be erased, as described below).**

4 **If you do not want to proceed with the erase operation, press any other key (such as the STOP key).**

NOTE

If only one track has been recorded on the disc, only the “erase disc” option (5.2.2, “Erasing a whole disc”) is available.

5 **When the range of tracks to be erased has been selected, press the MULTI JOG control or press the ENTER key ❽.**

NOTE

This operation cannot be undone. If you are going to erase a track or tracks, make sure that they are recordings that you really want to erase!

6 **If the erase operation has been selected, the word ERASE flashes on the display, and the timer starts counting down from 0:20 (20 seconds).**

7 **At the end of the countdown period, the word Complete appears briefly on the display, and the display returns to the track/time display.**

5.2.2 Erasing a whole disc

When you want to erase a whole disc (all the tracks on a disc), follow the procedure below:

1 **With the disc loaded, and the unit in stop mode, press the ERASE key (⑯ or ⓭).**

2 **Turn the MULTI JOG control ⑪ or use the SKIP keys ⓲ until the display shows ERASE DISC?.**

3 **If you do not want to proceed with the erase operation, press any other key (such as the STOP key).**

4 **To erase the contents of the disc, press the MULTI JOG control or press the ENTER key ❽.**

NOTE

This operation cannot be undone. If you are going to erase a disc, make sure that it contains only recordings that you really want to erase!

5 **If the erase operation has been selected, the word ERASE flashes on the display, and the timer starts counting down a time value shown on the display.**

6 **At the end of the countdown period, the word Complete appears briefly on the display, and the display returns to the track/time display.**

5.2.3 Refreshing a disc

If a CD-RW disc has been corrupted at any point (e.g. power has been cut off in the middle of a finalize operation, etc.), or has been used in a computer or another piece of equipment that makes it unreadable, it is possible to “refresh” the disc to make it usable again. This is equivalent to formatting a disk on a computer to prepare it for use. This operation destroys all data on the disc.

1 **With the disc loaded, and the unit in stop mode, press the ERASE key (⑯ or ⓭).**

2 **Turn the MULTI JOG control ⑪ or use the SKIP keys ⓲ until the display shows DISC REFRESH.**

3 **If you do not want to proceed with the refresh operation, press any other key (such as the STOP key).**

4 **To refresh the disc, press the MULTI JOG control or press the ENTER key ❽.**

NOTE

This operation cannot be undone. If you are going to refresh a disc, make sure that it contains only recordings that you really want to erase!

5 **If the erase operation has been selected, the word ERASE flashes on the display, and the timer starts counting down. The refresh operation typically takes around 20 minutes or so.**

6 **At the end of the countdown period, the word Complete appears briefly on the display, and the display returns to the track/time display (of course, there are no tracks, and the disc is as the start, with the total time of the disc available for recording).**

6 - Reference and specifications

6.1 Troubleshooting

Symptom	Possible cause and remedy
No sound on playback (the disc is playing and the time display is incrementing)	The connections from the CD-RW700 to the amplifier system have not been made properly, or the input source selector is not at the correct position.
No meter indication when a source is input	Analog and/or digital volume (4.1.1, “Signal volume”) is set too low. The signal is being input through an unselected input source (4.1, “Input selection”)
D-IN UNLOCK!	A digital input source has been selected, but the source is not connected or is not switched on.
Cannot record on a disc (pressing **RECORD** shows MONITOR)	A finalized disc has been inserted. Use an unfinalized recordable disc.
PGM Full !	An attempt has been made to create more than 25 program steps.
Disc Full! or Cannot Rec!	There is no space on the disc, or no tracks are available, or a general recording error has occurred
SCMS !!	You are trying to record digitally from a “second-generation” digital copy. Try analog recording (4.3.3, “SCMS and the CD-RW700”.
Not FS 44.1k!	You are trying to do digital direct recording from a source which has not been recorded at 44.1 kHz (4.3.4, “Digital direct”)
No Inc Mode	Auto track is not on when an attempt is being made to rehearse track split times.
Timer Error!	An attempt has been made to set timer recording (2.5.3, “Using the timer”) with a non-recordable disc, or no disc.
Disc Error	The disc is bad. If it is a CD-RW disc, you may refresh it (5.2.3, “Refreshing a disc”), otherwise replace the disc.
Erase Error!	An error has occurred while trying to erase the disc
The display suddenly shows PMA Writing	Either the disc is full (this message will be prefaced by Disc Full!! (4.2.5, “Full disc”) or an attempt has been mode to write more than 99 tracks.
Rec Error	General recording error. This may be caused by vibration or a shock during recording, for example, or may be the result of a bad disc. Try recording again with the same disc. If the message appears again, try a different disc.
No OPC Area!	Appears only with CD-R discs. The OPC area is full, and no further recording is possible on this disc. Finalize the disc.
OPC Area CLR	Seen when a CD-RW disc OPC area is full, and the CD-RW700 automatically cleans up the OPC area for further recording.
Need Repair!, System Err!, Tray Error!, Drive Error!	These messages indicate an internal error in the unit. Consult your dealer or distributor regarding repair facilities.

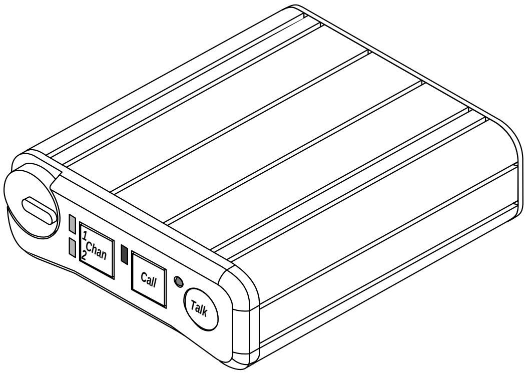
6.2 Specifications

Recording media type	CD-R, CD-RW
Recording resolution	16-bit linear
Recording sampling frequency	44.1 kHz
Frequency convertor input	32 kHz — 48 kHz
Frequency response	20 Hz — 20 kHz (playback ±0.5 dB, recording ±1 dB)
S/N ratio	98 dB (playback) 92 dB (recording)
Dynamic range	94 dB (playback) 92 dB (recording)
Total harmonic distortion	0.004% (playback) 0.005% (recording)
Channel separation	90 dB (playback : 1 kHz) 80 dB (recording : 1 kHz)
Wow & flutter	Unmeasurable (< 0.001%)
Analog inputs	Unbalanced RCA
Nominal input level	–10 dBV (FS –16 dB)
Maximum input level	+ 6dBV
Input impedance	33 kΩ (unbalanced)
Analog outputs	Unbalanced RCA
Nominal output level	–10 dBV (FS –16 dB)
Maximum output level	+6 dBV
Output impedance	800 Ω (unbalanced)
Headphone output	6 mm (1/4”) stereo
Output level	35 mW + 35 mW (into 32 Ω)
Digital inputs	
COAXIAL	RCA pin, IEC60958 TYPE II
OPTICAL	TOSLINK , IEC60958 TYPE II
Digital outputs	
COAXIAL	RCA pin, IEC60958 TYPE II
OPTICAL	TOSLINK , IEC60958 TYPE II
Voltage requirements	USA/Canada 120 VAC, 60 Hz U.K./Europe 230 VAC, 50 Hz Australia 240 VAC, 50 Hz General export 120/230 VAC, 60 Hz
Power consumption	27 W
Dimensions w x h x d	483 x 98 x 312 (mm) 19 x 3.9 x 12.3 (in)
Weight	6.3 kg (13.9 lbs)
Operating temperature	5°C to 35°C (41°F to 95°F)
Supplied accessories	RC-RW700 remote control unit 2 x type 3 (AA) batteries

PRINTED IN CHINA MA-0335

Telex

Operating Instructions


1
Chan
2
Call
Talk

BP-1002/BP-2002
Belt Packs
Audiocom® Intercom System

93507740-000 Rev E 10/2003

INTRODUCTION

The Audiocom® BP-1002 and BP-2002 are microprocessor controlled one- and two-channel intercom belt packs. An internal switch and jumper setting allows the units to be used with Clear-Com® components, if desired. Other internal switch and jumper settings allow the unit to be uniquely configured to the operator's requirements.

FCC STATEMENT

This equipment uses, and can radiate radio frequency energy that may cause interference to radio communications if not installed in accordance with this manual. The equipment has been tested and found to comply with the limits of a Class A computing device pursuant to Subpart B, Part 15 of FCC Rules which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area may cause interference which the user (at his own expense) will be required to correct.

CE

This product meets the Electromagnetic Compatibility Directive, 89/336/EEC.

®Audiocom is a registered trademark of Telex Communications, Inc., Burnsville, Minnesota 55337
®Clear-Com is a registered trademark of Clear-Com Intercom Systems

End-User License Agreement for Telex® Software

IMPORTANT - Please read this document carefully before using this product.

THIS DOCUMENT STATES THE TERMS AND CONDITIONS UPON WHICH TELEX COMMUNICATIONS, INC. (THE 'COMPANY') OFFERS TO LICENSE THE INSTALLED SOFTWARE OR PROGRAM (THE 'SOFTWARE') FOR USE WITH THE PRODUCT IN WHICH IT WAS INSTALLED. YOU ARE AGREEING TO BECOME BOUND BY THE TERMS OF THIS AGREEMENT. IF YOU DO NOT AGREE TO THE TERMS OF THIS AGREEMENT, DO NOT USE THIS PRODUCT. PROMPTLY RETURN THE PRODUCT TO THE PLACE WHERE YOU OBTAINED IT FOR A FULL REFUND.

The installed software as supplied by the Company is licensed, not sold, to you for use only under the terms of this license, and the Company reserves all rights not expressly granted to you. You own the product or other media on or in which the Software is originally or subsequently recorded or fixed, but the Company retains ownership of all copies of the Software itself.

1. License: This license allows you to use the Software for internal purposes only on a single product in which it was installed.
2. Restrictions: (a)You may not market, distribute or transfer copies of the Software to others or electronically transfer or duplicate the software. YOU MAY NOT REVERSE ENGINEER, DECOMPILE, DISASSEMBLE, MODIFY, ADAPT, TRANSLATE, RENT, LEASE OR LOAN THE SOFTWARE OR CREATE DERIVATIVE WORKS BASED ON THE SOFTWARE OR ANY ACCOMPANYING WRITTEN MATERIALS. (b) The Software and the accompanying written materials are copy-righted. Unauthorized copying of the Software, including portions thereof or the written materials, is expressly forbidden. (c) You understand that the Company may update or revise the Software and in so doing incurs no obligation to furnish such updates to you.
3. Limited Warranty: The Company does not warrant that the operation of the Software will meet your requirements or operate free from error. The Company DISCLAIMS ALL OTHER WARRANTIES AND CONDITIONS EITHER EXPRESS OR IMPLIED, INCLUDING THE WARRANTIES OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE AND NON-INFRINGEMENT OF THIRD PARTY RIGHTS.
4. Limited Liability: The liability of the Company for any claims arising out of this License based upon the Software, regardless of the form of action, shall not exceed the greater of the license fee for the Software

OPERATION

SYSTEM POWER

The BP-2002 belt pack receives power externally, in one of two ways:

- Via the intercom channel.
- Via the local-power (pin 2) of the intercom channel connector

The BP-1002 belt pack receives power externally, via the intercom channel.

Both the BP-1002 and BP-2002 will pass system power through to subsequent belt packs that are “daisy chained” together.

INITIAL BP-1002/BP-2002 SETUP

The channel termination is initially set for balanced operation, which is compatible with other Audiocom® equipment. If the unit is going to be connected to Clear-Com equipment, one switch and several jumpers must be changed as described in the section on Clear-Com Setup in this manual.

The headset microphone type is initially set for single-ended dynamic operation. To change the microphone type to balanced dynamic, refer to Table 2 for the position of JP9.

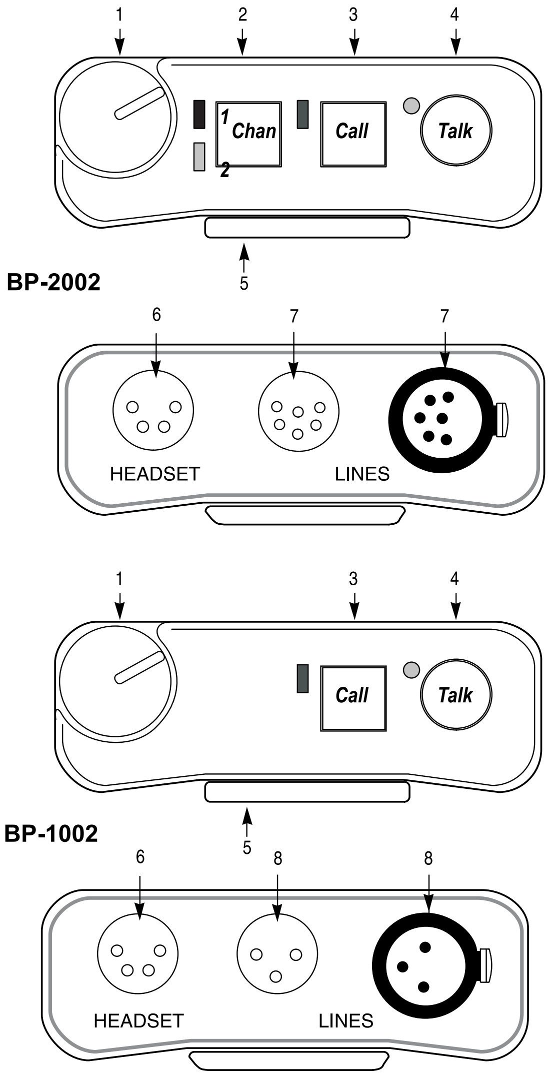

1
2
3
4
1
Chan
2
Call
Talk
5
BP-2002
6
7
7
HEADSET
LINES
1
3
4
Call
Talk
BP-1002
5
6
8
8
HEADSET
LINES

Figure 1. BP-2002 & BP-1002 Connections and Controls

EXTERNAL CONNECTIONS & CONTROLS

NOTE: The numbers refer to the callouts in Figure 1.

1. VOLUME CONTROL: Use this control to adjust the headset listen level.

2. CHAN BUTTON AND INDICATORS: The Chan button (only on the BP-2002) allows the user to select which intercom channel is active. The yellow indicator next to the 1 or 2 lights to show the active channel. Press the Chan button to change the channel selection, the yellow indicator for that channel will light.

3. CALL BUTTON AND INDICATOR: The Call function allows the user to send or receive signals to other devices on the intercom channel selected. The Call button operates in two ways:

   Call receive: When there is an incoming call signal, the indicator is red. (If Audible Call Alert is enabled, incoming calls will cause beeps in the headset.) On the BP-2002, calls can be received on the selected channel only.

   Call send: To send a call signal to all stations on a channel, press and hold the Call button until a verbal response is received. The indicator will glow red. On the BP-2002, calls can be sent on the selected channel only.

4. TALK BUTTON AND INDICATOR: The Talk button activates the headset microphone and operates in two ways:

Latched Mode:

Tap the button once to talk. The indicator will glow green. Tap the button again when finished with a conversation.

Momentary Mode:

Press and hold the button to talk momentarily. Release the button when finished talking.

NOTE: On the BP-2002, if no headset is connected when the Talk button is pressed, the Talk button will have the same function as the Chan button.

5. SIDETONE CONTROL: When using a headset, this control adjusts your own voice level heard in the head phones. To adjust the level, tap the Talk button once to turn on the headset microphone. Then, use a small flat-blade screwdriver to increase or decrease your voice level while talking into themicrophone. (This control is accessible by removing one screw of belt clip.)

6. HEADSET CONNECTOR: This connector accepts a four-wire Telex® boom-microphone headset.

7. INTERCOM CHANNEL CONNECTORS: On the BP-2002, intercom channels are connected via a pair of 6-pin connectors (one male and one female). The male and female connectors are wired together in parallel, providing a “loop through” at each connector pin. Use one connector to connect to the intercom channel. Use the other connector to “daisy chain” a cable to the next belt pack or other station on the channel.

Local Power Input (BP-2002 only)
Normally the BP-2002 is powered from the intercom system and will turn on with the intercom system. The BP-2002 belt pack may also be powered from an optional power supply (18-30 VDC) connected between pin 2 (+) and pin 1 (-) of the intercom channel connector.

8. INTERCOM CHANNEL CONNECTORS: On the BP-1002, the intercom channel is connected via a pair of 3-pin connectors (one male and one female). The male and female connectors are wired together in parallel, providing a “loop through” at each connector pin. Use one connector to connect to the intercom channel. Use the other connector to “daisy chain” a cable to the next belt pack or other station on the channel.

OPERATING MODES

The microprocessor within the BP-1002/BP-2002 controls four modes of operation that affects the Microphone Kill and Audible Call Alert features.

MODE (BEEPS)	MIC KILL	AUDIBLE CALL ALERT
1	Disabled	Disabled
2 (Default)	Enabled	Disabled
3	Disabled	Enabled
4	Enabled	Enabled

Table 1. Operating Modes

Changing Modes of Operation:

Perform the following steps to change the mode of operation.

1. Both the Talk and Call indicators should be off.

2. Press and hold the Talk key, then press and hold the Call key, then release both keys. The Call indicator should now glow red.
   (The number of beeps heard in the headset indicates the current mode of operation.)

3. Press the Call key to change to the next mode of operation. Each press of the Call key will cause the BP-1002/BP-2002 to change to the next mode of operation.

4. When the desired mode is reached, press the Talk key to select that mode and exit the mode changing function. (Beeps will be heard in the headset when the mode changing function is exited. The number of beeps heard indicates the selected mode of operation.)

NOTE: Each time the intercom system power is turned on, the beltpack will reset to the default mode of operation (Mode 2).

INTERNAL SWITCHES, JUMPERS AND ADJUSTMENTS

There are several internal switches, jumpers and an adjustment that affect operation. These are described below. To gain access to the switches, jumpers and the adjustment, disconnect all power and line connections. Remove two screws from the top of each side and two screws from bottom of each side. Switch, jumper and adjustment locations are shown in Figure 2.

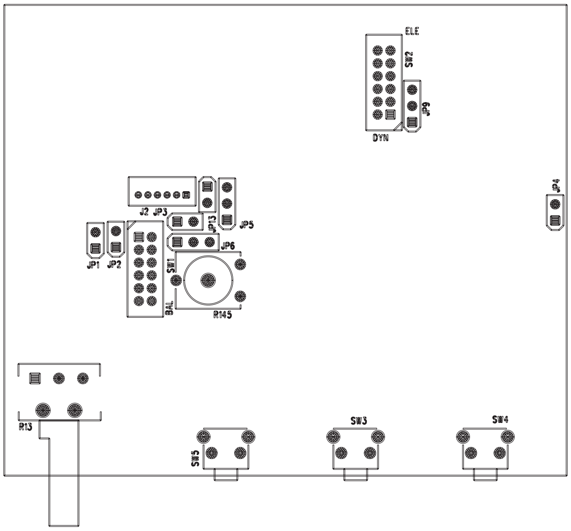


Figure 2. Internal Switches, Jumpers and Adjustments. Note, if you have 9350-7749-000 board, figure 4 on page 19.

> **NOTE:** Figure 2 shows a BP-2002 with the switches and jumpers in their factory default positions (referenced in Table 2).

The SIDETONE adjustment is also accessible behind the screw that holds the belt clip (callout 5 in Figure 1).

The functions of the internal switches and jumpers are described in Table 3.

NOTE: On the BP-1002, jumper JP6 must always have pins 2 and 3 shorted.

Side Tone Adjustment (R145)

The side tone adjustment is accessible either internally (refer to Figure 2) or by removing the belt clip mounting screw (callout 5 in Figure 1).

To adjust the level of your own voice heard in the headphones, tap the Talk button once to turn on the headset microphone. Then, use a small flat-blade screwdriver to increase or decrease your voice level while talking into the microphone.

Clear-Com Setup

Make the following switch and jumper changes when the belt pack is used with Clear-Com equipment:

BP-2002

1. SW1 must be placed in the UNBAL position.
2. JP4 must have pins 1 and 2 shorted (U position).
3. JP5 jumper must be removed and placed on JP2.
4. JP2 must have pins 1 and 2 shorted (using jumper from JP5).
5. JP1 must have pins 1 and 2 shorted.

BP-1002

1. SW1 must be placed in the UNBAL position.
2. JP4 must have pins 1 and 2 shorted (U position).
3. JP6, JP5 must have pins 2 and 3 shorted at all times.

Product	Default Jumper Settings
BP-1002	Shorting across Pins 1 and 2: JP13 Shorting across Pins 2 and 3: JP5, JP6 and JP9 Not Populated: JP1, JP2, and JP3 Populated on Pin 1 Only: JP4
BP-2002	Shorting across Pins 1 and 2: JP9 and JP13 Shorting across Pins 2 and 3: JP5 and JP6 Not Populated: JP3 Populated on Pin 1 Only: JP1, JP2, and JP4

***Table 2. Default Settings**. If you are using the 000 board, see page 19 for the default settings.*

Table 3. *Internal Switches and Jumpers*

Jumper/Switch Number	Jumper or Switch Function	Default Setting
SW3 ^	Electret/Dynamic Microphone Select (Must be set to Dynamic when JP9 functions are designed	Dynamic
JP9 ^ ^	Balanced/Single-ended Dynamic Mic (SW1 must be set to Dynamic) Single-ended Mic: Pins 1&2 shorted	Pins 1&2 shorted
^	Balanced Mic: Pins 2&3 shorted	
JP6 ^ ^	Power Select Channel One Power: Pins 2&3 shorted (On BP-1002, pins 2&3 always shorted)	Pins 2&3 shorted
^	Channel Two Power: Pins 1&2 shorted	
JP5	Power Select	Pins 2&3 shorted
^ ^	Channel One Power: Pins 2&3 shorted (On BP-1002, Pins 2&3 always shorted.) Channel Two Power: Pins 1&2 Shorted	
^ ^	(Must be set for same channel as JP6 for proper power operation in Audiocom® mode on BP-2002) (Must be used in conjunction with SW1)	
SW1	Clear-Com / Audiocom® Operation	Balanced
^ ^	Unbalanced / Balanced Line (Must be used in conjunction with JP4, JP5, JP2, and JP1)	
JP4	Clear-Com / Audiocom® Operation Pins 1&2 shorted: UNBAL (Clear-Com)	Jumper on one pin only
JP2 ^	DC Call Detect (Clear-Com operation only, not used in Audiocom® mode) BP-2002 only Disabled: No pins jumpered	No jumper present
^ ^	Enabled: Pins 1&2 shorted (To enable, use jumper from JP5)	
JP1 ^ ^	DC Call Detect (Clear-Com operation only, not used in Audiocom® mode) BP-2002 only Disabled: No pins jumpered Enabled: Pins 1&2 shorted	Jumper on one pin only
JP13	For RTS™ use only	Must be installed

Note: In case of local power use, install a jumper on pins 2 & 3 of U6

CONNECTOR PIN CONFIGURATIONS

Headset Connector

Type: XLR-4M (callout 6 in Figure 1)

Pin 1	Headset microphone low
Pin 2	Headset microphone high
Pin 3	Headphone high
Pin 4	Headphone low

Intercom Channel Connectors

BP-1002

Type: One XLR-3M and XLR-3F pair (callout 8 in Figure 1)

Audiocom® Mode (Internal switch SW1 and jumpers JP4, JP5, JP2 and JP1 set to BAL position)

Pin 1	Common
Pin 2	Intercom audio low and +24 VDC input
Pin 3	Intercom audio high and +24 VDC input

Clear-Com Mode (Internal switch SW1 and jumpers JP4, JP5, JP2 and JP1 set to UNBAL position)

Pin 1	Common
Pin 2	+30 VDC input
Pin 3	Intercom audio/call signal

BP-2002

Type: One XLR-6M and XLR-6F pair (callout 7 in Figure 1)

Audiocom® Mode (Internal switch SW1 and jumpers JP4, JP5, JP2 and JP1 set to BAL position)

Pin 1	Common
Pin 2	Local power (+24 VDC)
Pin 3	Channel A intercom audio low and +24 VDC input
Pin 4	Channel A intercom audio high and +24 VDC input
Pin 5	Channel B intercom audio low and +24 VDC input
Pin 6	Channel B intercom audio high and +24 VDC input

Clear-Com Mode (Internal switch SW1 and jumpers JP4, JP5, JP2 and JP1 set to UNBAL position)

Pin 1	Common
Pin 2	Local power (14 to 30 VDC)
Pin 3	Channel A +30 VDC input
Pin 4	Channel A intercom audio/call signal
Pin 5	Channel B +30 VDC input
Pin 6	Channel B intercom audio/call signal

SPECIFICATIONS

GENERAL:

Power Requirements:
Channel supplied: 24 VDC nominal, 45 to 70 mA
Local-power: (BP-2000 only) 24 VDC nominal (18 to 30 VDC), 45 to 70 mA

Environmental Requirements:
Storage: -20ºC to 80ºC; 0% to 95% humidity, non-condensing
Operating: -15ºC to 60ºC; 0% to 95% humidity, non-condensing

Dimensions: 5.0" (127 mm) H x 3.5" (88.9 mm) W x 1.8" (45.7 mm) D

Weight: 1.5 pounds (0.68 kg)

INTERFACE REQUIREMENTS:

Headset:
50 to 200 ohm dynamic microphone
150 to 600 ohm headphones

Balanced Intercom Channel:
Output Level: 1 Vrms nominal
Input Impedance: 300 ohm ±5%
Bridging Impedance: greater than 10,000 ohm
Call Signalling:
Send: 20 kHz ±100 Hz, 0.5 Vrms ±10%
Receive: 20 kHz ±800 Hz, 100 mVrms
Mic-Off Frequency:
Detect: 24 kHz ±800 Hz, 100 mVrms
Noise Contribution: less than -60 dB on the line

Unbalanced Intercom Channel:
- Output Level: 480 mVrms ±10%
- Input Impedance: 200 ohm ±5%
- Bridging Impedance: greater than 10,000 ohm
- Call Signalling:
  - Send: 12 ±3 VDC
  - Receive: 4 VDC minimum

HEADPHONE AMPLIFIER:

Voltage Gain:
20 ±3 dB from the line

Maximum Output:
60 mW into 150 ohm

Frequency Response:
200 Hz to 8 kHz with no more than 4 dB deviation

Audible Alert:
1 kHz, at the headset

Total Harmonic Distortion:
Less than 2% at 50 mW from the line

Sidetone:
20 dB minimum range, adjustable

DYNAMIC MICROPHONE AMPLIFIER:

Voltage Gain:
Mic to Channel; 45 ±3 dB, before limiting

Mic to Headphone; adjustable, 65 dB range into 150 ohm

Frequency Response:
200 Hz to 8 kHz +1/-3dB

Total Harmonic Distortion:
Less than 1% at Channel output at 1 kHz

ELECTRET MICROPHONE AMPLIFIER:
Voltage Gain:
Mic to Channel; 24 ±3 dB, before limiting

Mic to Headphone; adjustable, 45 dB ±10% into 150ohm

Frequency Response:
200 Hz to 8 kHz +1/-3dB

Total Harmonic Distortion:
Less than 1% at Channel output

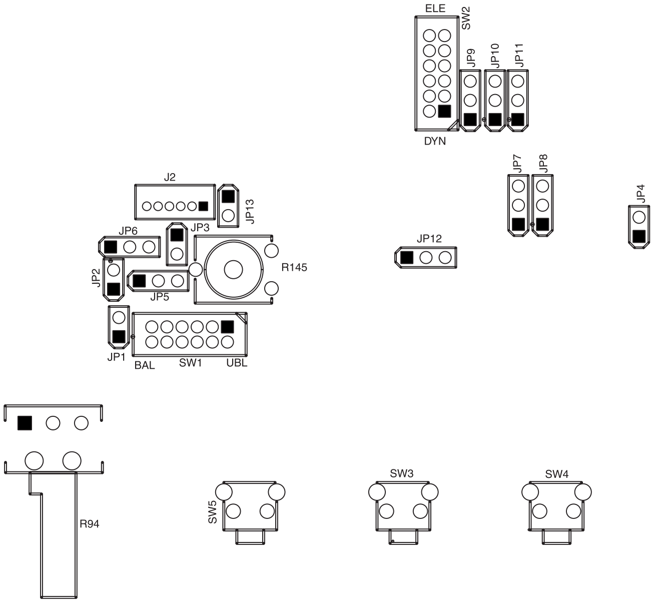

ELE
SW2
JP9
JP10
JP11
DYN
J2
JP13
JP7
JP8
JP4
JP6
JP3
JP12
R145
JP2
JP5
JP1
BAL
SW1
UBL
SW3
SW4
SW5
R94

Figure 4. ***Board Number 9040-7740-000.***

Product	Default Jumper Settings
BP-1002	Shorting across Pins 1 and 2: JP7, JP8, JP10, JP11, JP12, and JP13 Shorting across Pins 2 and 3: JP5, JP6 and JP9 Not Populated: JP1, JP2, and JP3 Populated on Pin 1 Only: JP4
BP-2002	Shorting across Pins 1 and 2: JP7, JP8, JP9, JP10, JP11, JP12, and JP13 Shorting across Pins 2 and 3: JP5 and JP6 Not Populated: JP3 Populated on Pin 1 Only: JP1, JP2, and JP4

Default Settings

APPENDIX A
OPTIONAL FOOTBALL MODIFICATIONS

GENERAL DESCRIPTION

In Audiocom® Intercom Systems, the Mic-Kill feature is used to turn off any activated microphones on a selected channel. The Mic-Kill feature is activated when the beltpack receives a 24 kHz signal from the channel. In some applications the Mic-Kill and Call signal features are not desired and need to be disabled.

ELIMINATING MIC-KILL AND CALL SIGNAL

CAUTION

TO PREVENT DAMAGE TO THE EQUIPMENT, THESE MODIFICATIONS SHOULD ONLY BE MADE BY QUALIFIED TECHNICIANS.

If desired, the Mic-Kill and Call signal features can be disabled in the BP-2002/BP-1002 by removing a surface-mounted capacitor from the printed circuit board. Perform the following:

1. Before making changes to the printed circuit board, disconnect all power and line connections from the beltpack.

2. Refer to Figure 3 and locate C61 on the underside of the printed circuit board.

3. Remove capacitor C61 from the printed circuit board.

RESTORING MIC-KILL AND CALL SIGNAL

To restore the Mic-Kill and Call signal features, replace C61 with Telex® part number 102879-218, 1500 pF, 50V capacitor.

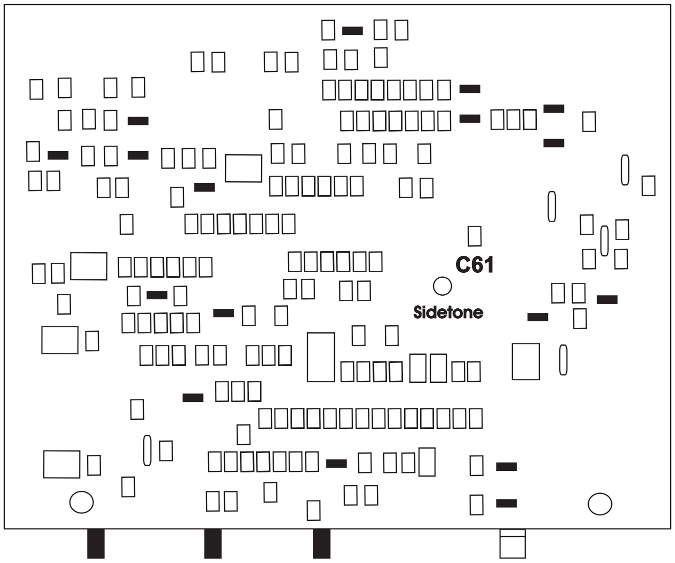

C61
Sidetone

Figure 3. Printed Circuit Board

FACTORY SERVICE

All equipment returned for repair must be accompanied by documentation stating your return address, telephone number and proof of date of purchase, along with a description of the problem. In lieu of this, you may obtain a Return Authorization form from our Customer Service Department.

Customer Service Department
Telex Communications, Inc.
12000 Portland Avenue South
Burnsville, Minnesota 55337 U.S.A.
Telephone: (800) 392-3497
Fax: (800) 323-0498

Return equipment to:

Service Department
Telex Communications, Inc.
West 1st Street
Blue Earth, Minnesota 56013 U.S.A.

WARRANTY REPAIRS - If in warranty, no charge will be made for the repairs. Equipment being returned for warranty repair must be sent prepaid and will be returned prepaid.

NON-WARRANTY REPAIRS - Equipment that is not under warranty must be sent prepaid to Telex. If requested, an estimate of repair costs will be issued prior to service. Once your approval for repair, and repair of equipment is completed, the equipment will be returned on a collect basis. Collect charges may be avoided by sending a signed check for payment in full along with your signed estimate approval form (the estimate includes the shipping charge).

TELEX®

TELEX COMMUNICATIONS, INC. • 12000 Portland Avenue South, Burnsville, MN 55337 USA

Telex® CE

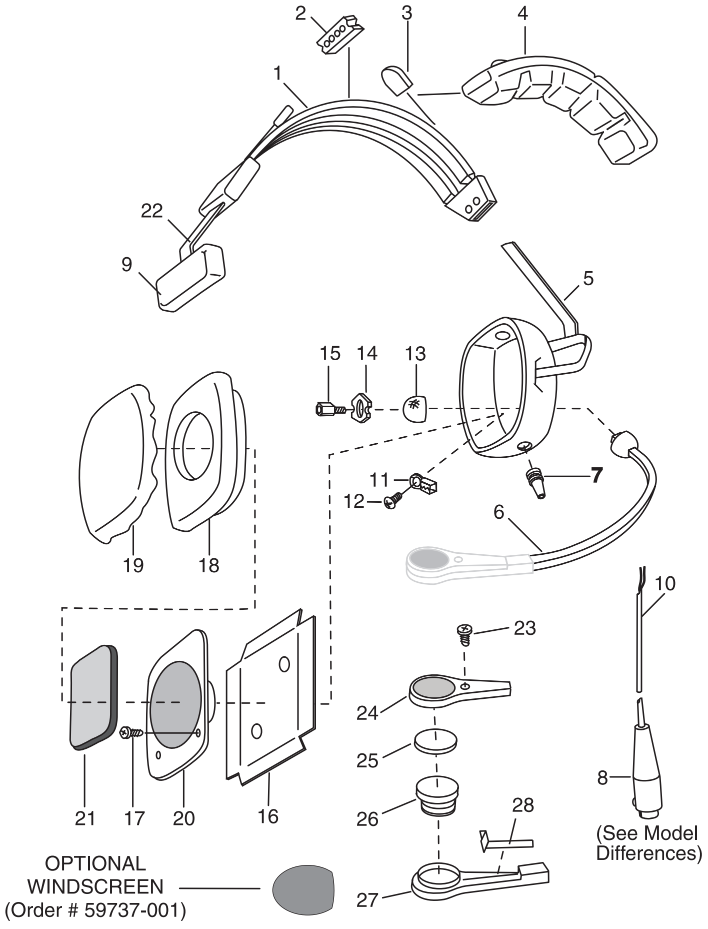
Parts List and Wiring Diagrams

Model PH-1 Headset (64438 Series)

The PH-1 headset is a medium weight, full cushion, single sided headset for the ultimate in day-long comfort. The PH-1 features a high quality monaural dynamic earphone with dynamic noise cancelling microphone on adjustable ball joint boom that can be positioned on either side of the head. The unit has a straight cord and is available in various connector configurations.

Specifications

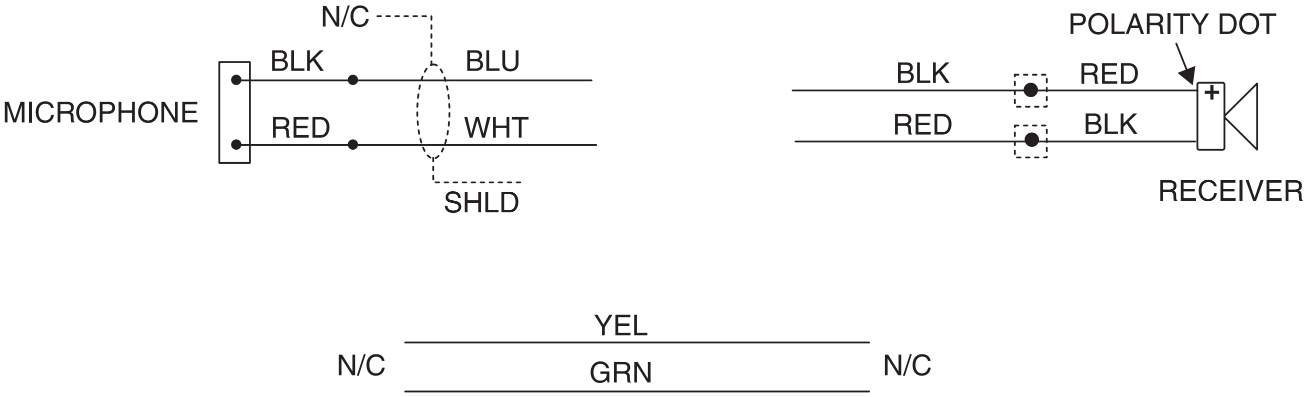
Receiver Type	Dynamic
Impedance	150 ohms
Microphone Type	Dynamic
Impedance	150 ohms
Weight	12 ozs
Cord	6' straight



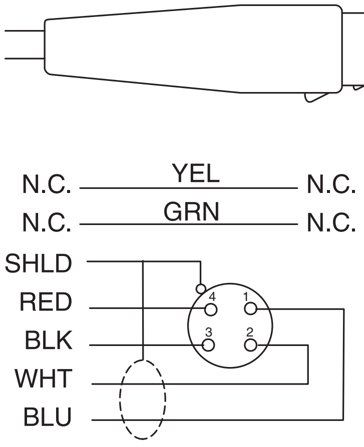
Parts List

ITEM NO.	DESCRIPTION	QTY	PART NO.
1	Headband Assy.	1	61162-046
2	Headband Spacer	1	63709-001
3	Slider Tip	1	61166-005
4	Headband	1	63801-001
5	Earcup-Yoke-Slider Assy	1	70398-034
6	Boom Assy	1	70398-001
7	Strain Relief	1	63465-006
8	Connector, PH-1 XLR-4F	1	35320-004
	Connector, PH-1R, XLR-4M	1	35398-004
	Connector, PH-1R5, XLR-5M	1	35398-005
9	Cushion	1	64442-000
10	Cord Unit	1	63988-063
11	Terminal	1	54462-001
12	Screw, No. 6-20 x 1/4	1	51856-008
13	Ball Half	1	63451-003
14	Ball Joint Spring	1	63453-000
15	Hex Spacer	2	57907-000
16	Foam Liner	1	63763-000
17	Screw, No. 4-40 x 1/4	2	51845-038
18	Ear Cushion	1	63444-001
19	Ear Cushion Cover	1	35772-000
20	Receiver Element/Earplate Assy	1	70398-035
21	Foam	1	70608-000
22	Slider	1	64440-001
23	Screw, No. 2-28 x 3/8	1	51856-002
24	Mic Housing Cover	1	63460-005
25	Foam	1	63124-018
26	Mic Element Assy	1	61585-037
27	Mic Housing	1	63459-005
28	Contact Tension Spring	1	63455-000

Wiring Diagram

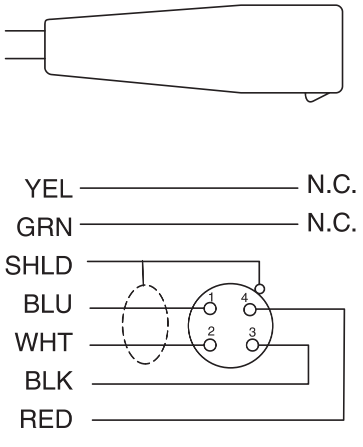

N/C
BLK
BLU
MICROPHONE
RED
WHT
SHLD
POLARITY DOT
BLK
RED
RED
BLK
RECEIVER
YEL
N/C
GRN
N/C

Connector Wiring


N.C.
YEL
N.C.
N.C.
GRN
N.C.
SHLD
RED
BLK
WHT
BLU
4
1
3
2

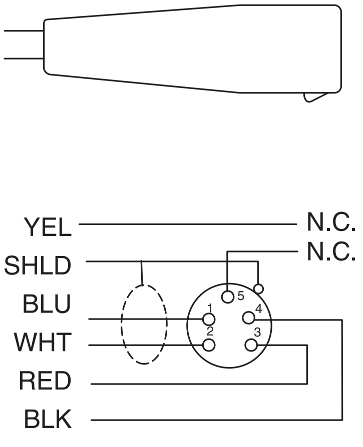
64438-100

PH-1


YEL
N.C.
GRN
N.C.
SHLD
BLU
WHT
BLK
RED
1
4
2
3

64438-102

PH-1R


YEL
N.C.
N.C.
SHLD
BLU
WHT
RED
BLK
5
1
4
2
3

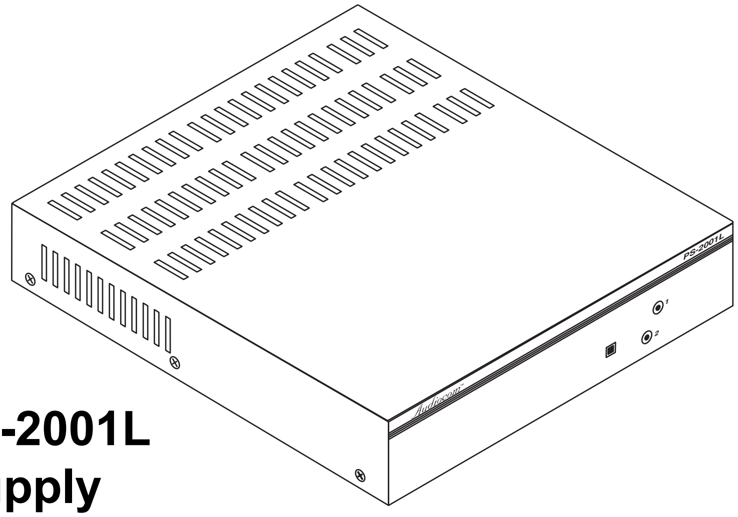
64438-103

PH-1R5

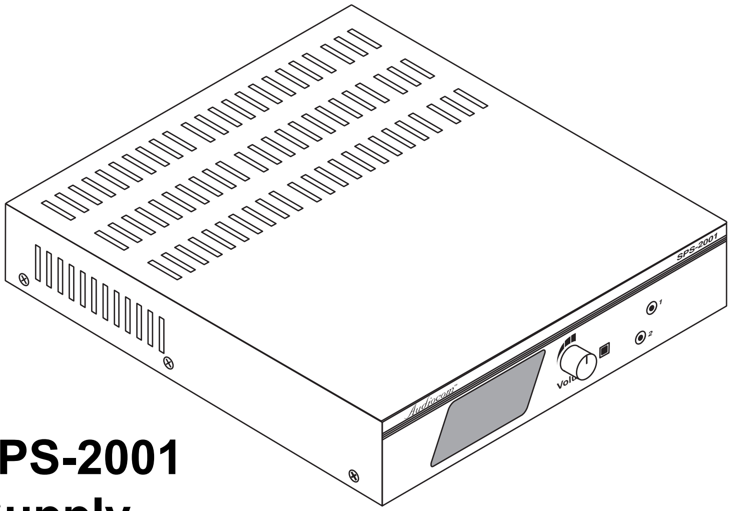
TELEX®

Telex®

User Instructions


PS-2001L
1
2
Audiocom

Model PS-2001L Power Supply


SPS-2001
1
2
Vol
Audiocom

Model SPS-2001 Power Supply

Audiocom® Intercom Systems

9350-7699-000 Rev. A, 4/2001

FCC Statement

This equipment uses, and can radiate radio frequency energy that may cause interference to radio communications if not installed in accordance with this manual. The equipment has been tested and found to comply with the limits of a Class A computing device pursuant to Subpart J, Part 15 of FCC Rules which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area may cause interference which the user (at his own expense) will be required to correct.

CE This product meets Electromagnetic Compatibility Directive 89/336/EEC

Table of Contents

Description 4

PS-2001L 4

SPS-2001 5

Installation 6

Unpacking 6

PS-2001L 6

SPS-2001 6

Configuration Switches 6

Intercom Channel and Program Connections 6

Desktop or Rackmount Installation 6

Specifications 12

General 12

Intercom Channels 12

General 12

Balanced Mode (set to BAL position) 12

Unbalanced Mode (set to UNBAL position) 12

Speaker Inputs (SPS-2001 Only) 12

Approvals 12

Power-Up Check 12

Description

The PS-2001L and SPS-2001 are versatile power supplies that can be used in a variety of Audiocom® intercom system applications. They both directly accept any AC input power from 100 to 240 VAC, 50/60 Hz and can be configured to power two separate intercom channels or one large intercom channel with twice the intercom station capacity. Additionally, the units can be configured for compatibility with Clear-Com* Intercom Systems. The units may be rack mounted or used on a desktop. For rack mounting, optional hardware is required. For desktop use, four non-marring rubber feet are supplied. The SPS-2001 also provides a speaker with two mixed inputs making it ideal for situations calling for a master station with a microphone / speaker combination.

PS-2001L

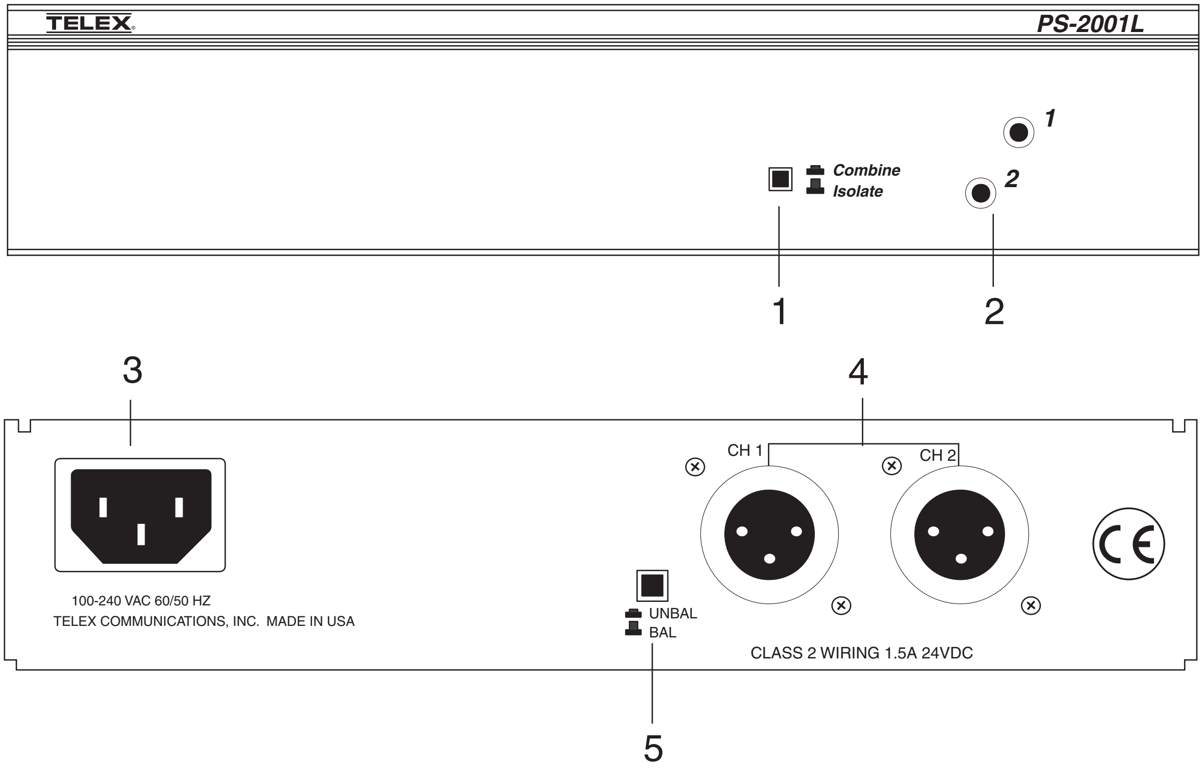


***Figure 1** - PS-2001L Front and Rear Panel Features.*

1. Combine / Isolate Switch: When in the Combine position, the unit will combine the audio signals of the two channels to create a single audio channel where all users can intercommunicate. When in the Isolate position, the unit will create two groups of completely independent users.

2. Channel Status Indicators: The indicators are green for normal operation and red when there is an overload or short circuit. The circuitry in the unit will automatically reset when the overload or short circuit is located and fixed.

3. Universal AC Power Input: The unit accepts any input power in the range of 100-240 VAC, 50/60 Hz.

4. Intercom Channel Connectors: When the Combine / Isolate switch is set to the Isolate position, each channel connector is powered separately and has completely separate intercom audio. When the Combine / Isolate switch is set to the Combine position, each channel is still powered separately, but the audio signals are combined so that all users on both channels can intercommunicate.

5. BAL / UNBAL Selector Switch: This selector switch allows the user to configure the unit for use in either an Audiocom® (BALANCED) or Clear-Com (UNBALANCED) system. Compatibility includes channel connector pin-outs, channel power requirements, and call signaling requirements. The default setting for this switch is in the Audiocom® (BAL) position.

SPS-2001

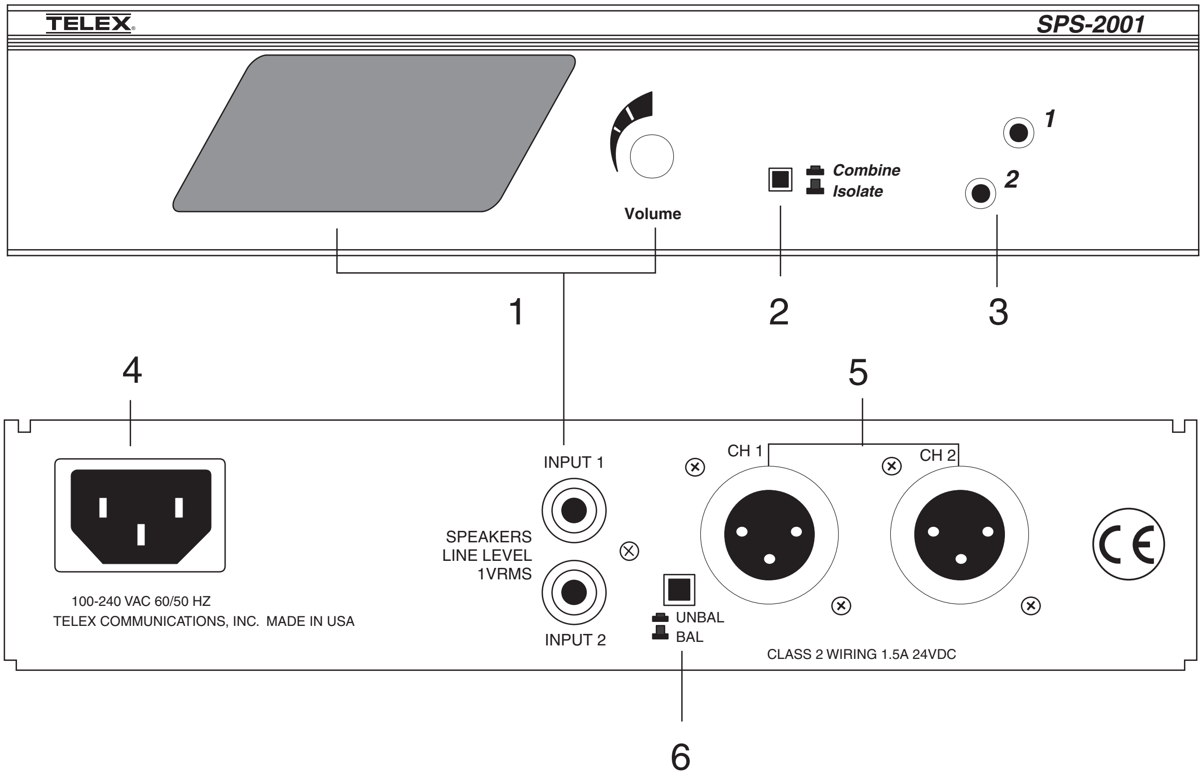


Figure 2 *- SPS-2001 Front and Rear Panel Features.*

1. Speaker, Inputs, and Volume Control: Two audio inputs are provided. The inputs are combined internally and sent as a monaural signal to the internal speaker amplifier. The Volume control adjusts the level to the front panel speaker. Typically, these inputs are used with the speaker outputs of a master station when the master station is set for use with a panel microphone and speaker instead of a headset.

2. Combine / Isolate Switch: When in the Combine position, the unit will combine the audio signals of the two channels to create a single audio channel where all users can intercommunicate. When in the Isolate position, the unit will create two groups of completely independent users.

3. Channel Status Indicators: The indicators are green for normal operation and red when there is an overload or short circuit. The circuitry in the unit will automatically reset when the overload or short circuit is located and fixed.

4. Universal AC Power Input: The unit accepts any input power in the range of 100-240 VAC, 50/60 Hz.

5. Intercom Channel Connectors: When the Combine / Isolate switch is set to the Isolate position, each channel connector is powered separately and has completely separate intercom audio. When the Combine / Isolate switch is set to the Combine position, each channel is still powered separately, but the audio signals are combined so that all users on both channels can intercommunicate.

6. BAL / UNBAL Selector Switch: This selector switch allows the user to configure the unit for use in either an Audiocom® (BALANCED) or Clear-Com (UNBALANCED) system. Compatibility includes channel connector pin-outs, channel power requirements, and call signaling requirements. The default setting for this switch is in the Audiocom® (BAL) position.

Installation

Unpacking

The package contains the following items. Contact the shipper or your Audiocom® dealer immediately if anything is damaged or missing. Detach and fill out the registration card and return it to Telex to properly register the product.

PS-2001L

Qty	Description
1	PS-2001L Power Supply
1	Warranty and registration card
1	User Manual
4	Rubber Feet (install for desktop use of the PS-2001L)
1	Power Cord

SPS-2001

Qty	Description
1	PS-2001L Power Supply
1	Warranty and registration card
1	User Manual
4	Rubber Feet (install for desktop use of the PS-2001L)
1	Power Cord
1	RCA to RCA Patch Cord

Configuration Switches

The **BAL / UNBAL** switch (located on the rear of the unit) is set to the Balanced (Audiocom) position when the unit is shipped from the factory. To set the switch to the Unbalanced (Clear-Com) mode, use a pointed object such as a pen to push in the switch.

The **Combine / Isolate** switch (located on the front of the unit) can be changed at any time by using a pointed object such as a pen to push in the switch. When the switch is in the Combine position, all users on both channels may intercommunicate. When the switch is set to the Isolate position, channels 1 and 2 cannot intercommunicate.

Intercom Channel and Program Connections

Channel Capacity: When connecting intercom stations to the PS-2001L or SPS-2001, determine the total current for all stations on each channel. The total per channel should not exceed 2A. If you exceed this limit, an overload indication will be provided and the output of the power supply will be turned off. Once the overload is corrected, the overload indication will disappear and the power supply output will be turned on. Note that if you are using DC wallpacks with some intercom stations, you do not need to add the current consumption of those stations to the total current.

Audiocom Connections: Intercom Cable wiring details are shown in Figure 3. Speaker interconnection cables (SPS-2001 only) are standard RCA phono types. For program input cable wiring, refer to your master intercom station user manual. Some typical intercom applications are shown in Figures 4 through 8.

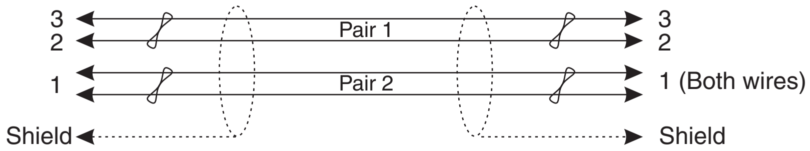
Clear-Com Applications: Refer to your Clear-Com system documentation for intercom cable wiring and connection details.

Desktop or Rackmount Installation

For desktop use, install the four supplied rubber feet to the bottom of the unit. Do not obstruct the cooling vents when using the unit on a desktop.

For rackmounting, use an optional Audiocom Rackmount Kit. Several kits are available to meet the requirements of your particular system. Follow the assembly instructions supplied with the rackmount kit.

TYPICAL 1-CHANNEL CABLE WIRING


3
2
1
Shield
Pair 1
Pair 2
3
2
1 (Both wires)
Shield

Cable Type: 22AWG Stranded, 2-Pair Twisted-wire, with Shield

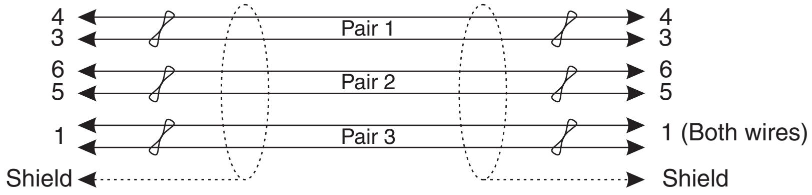
Connector Type: 3-Pin XLR Audio (Neutrik or Switchcraft)*

Pin 1: Common
Pin 2: Channel Audio / Power
Pin 3: Channel Audio / Power
Shield: Earth ground

Denotes twisted pair.

Denotes shield.

TYPICAL 2-CHANNEL CABLE WIRING


4
3
6
5
1
Shield
Pair 1
Pair 2
Pair 3
4
3
6
5
1 (Both wires)
Shield

Cable Type: 22AWG Stranded, 3-Pair Twisted-wire, with Shield

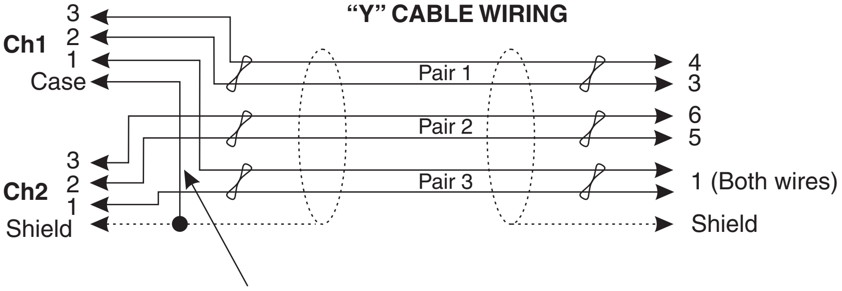
Connector Type: 6-Pin XLR Audio (Neutrik only, not compatible with 6-pin Switchcraft)*

Pin 1: Channel 1 & 2 Common
Pin 2: No connection
Pin 3: Channel 1 Audio / Power
Pin 4: Channel 1 Audio / Power
Pin 5: Channel 2 Audio / Power
Pin 6: Channel 2 Audio / Power
Shield: Earth ground

Denotes twisted pair.

Denotes shield.

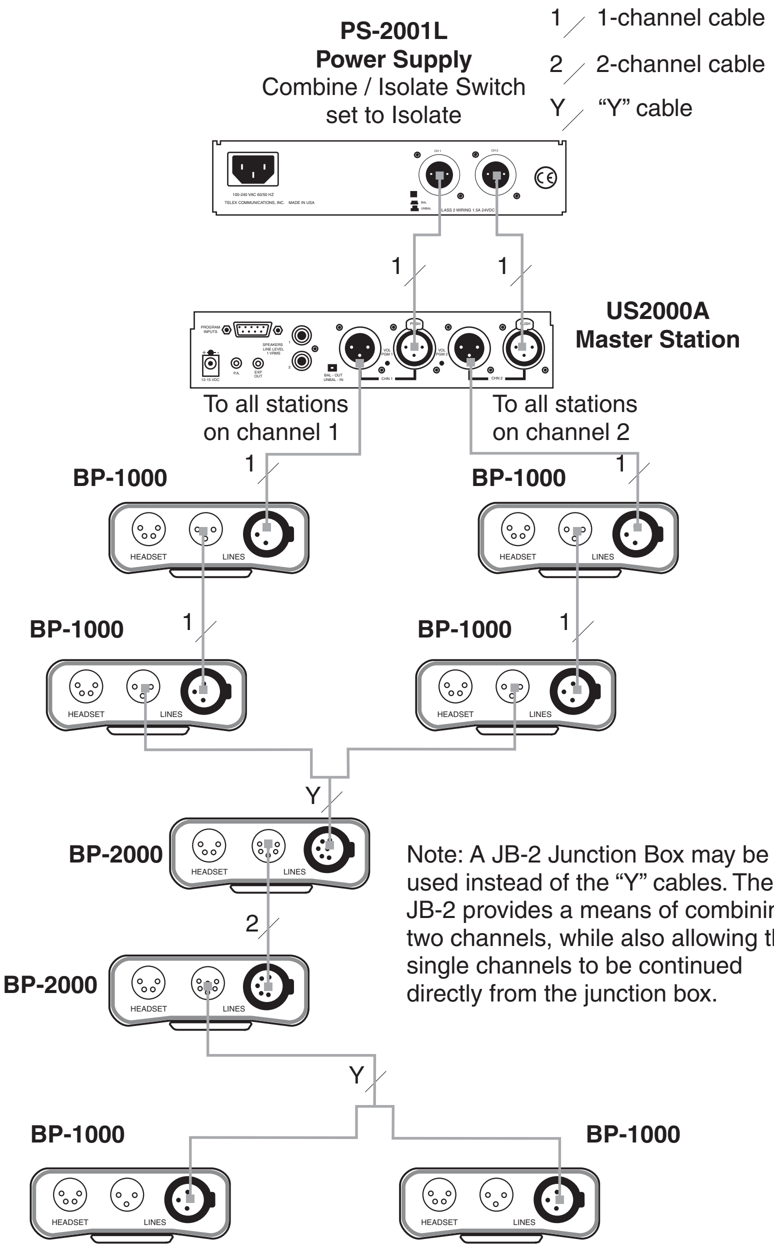
"Y" CABLE WIRING


Ch1
3
2
1
Case
Ch2
3
2
1
Shield
Pair 1
Pair 2
Pair 3
4
3
6
5
1 (Both wires)
Shield

Use second drain wire if available, or add an extra section of wire.

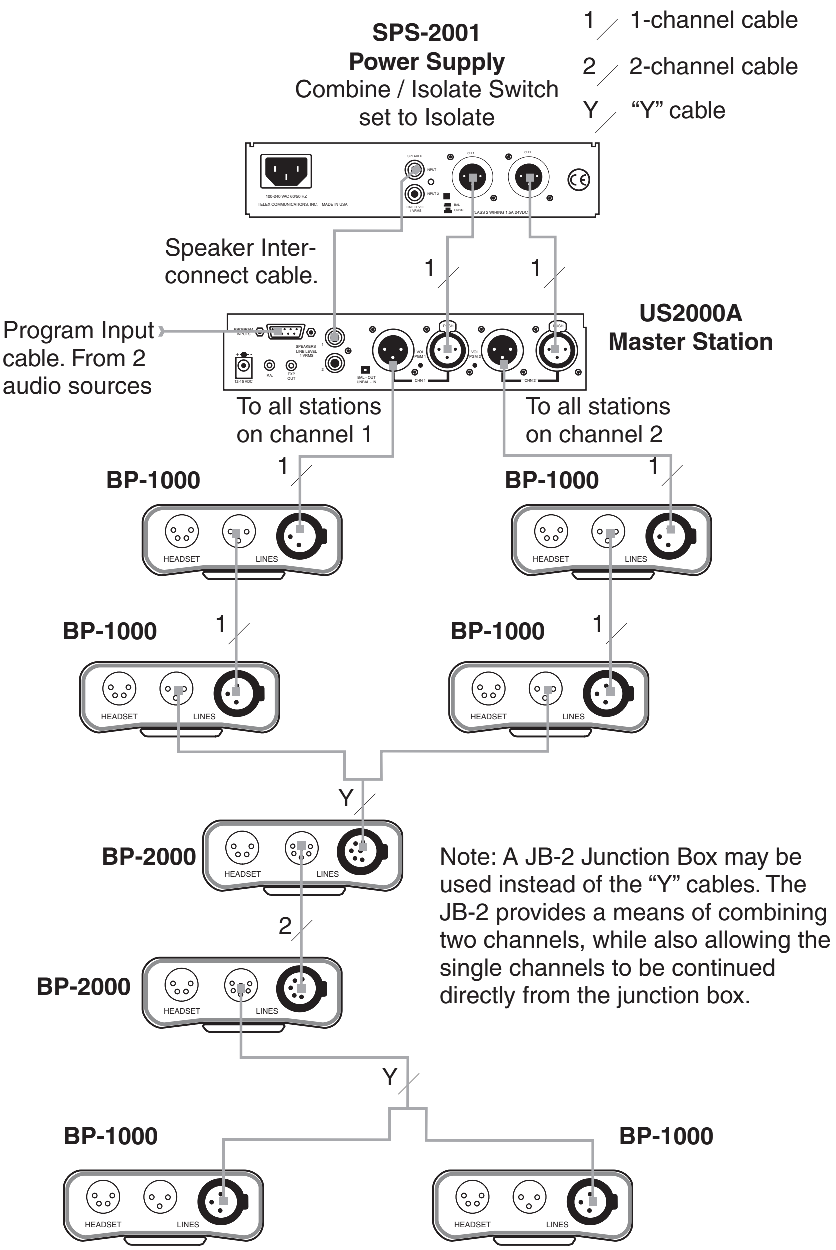
* Standard cables are generally constructed using a male connector at one end and a female connector at the other end. This allows several cables to be interconnected to create longer cable runs. Audiocom master stations, speaker stations and belt packs also typically provide both a male and female Neutrik connector for each intercom channel. This permits loop-through connection of several intercom stations using the standard cables. Audiocom power supplies use a 3-pin male Neutrik connector for each channel. Audiocom wallplates use male Neutrik connectors.

Figure 3 *- Cable Wiring Diagrams for Audiocom Applications.*


PS-2001L
Power Supply
Combine / Isolate Switch
set to Isolate
1 1-channel cable
2 2-channel cable
Y "Y" cable
1
1
US2000A
Master Station
To all stations
on channel 1
To all stations
on channel 2
1
1
BP-1000
BP-1000
HEADSET
LINES
BP-1000
BP-1000
1
1
Y
BP-2000
2
BP-2000
Y
BP-1000
BP-1000

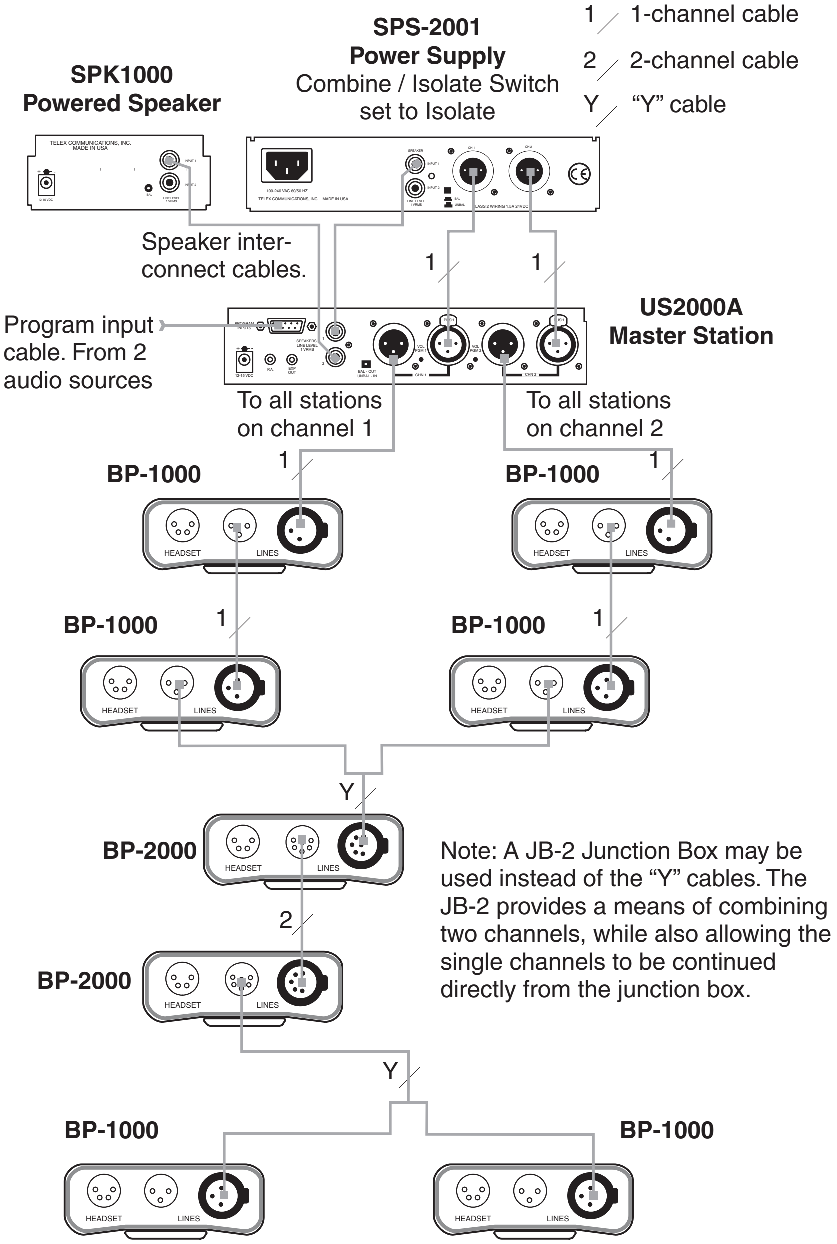
Note: A JB-2 Junction Box may be used instead of the "Y" cables. The JB-2 provides a means of combining two channels, while also allowing the single channels to be continued directly from the junction box.

Figure 4 *- Typical connections using a single PS-2001L Power Supply to power two channels.*


SPS-2001
Power Supply
Combine / Isolate Switch
set to Isolate
1 1-channel cable
2 2-channel cable
Y "Y" cable
Speaker Inter-
connect cable.
1
1
Program Input
cable. From 2
audio sources
US2000A
Master Station
To all stations
on channel 1
To all stations
on channel 2
BP-1000
1
BP-1000
1
HEADSET
LINES
BP-1000
1
BP-1000
1
Y
BP-2000
2
BP-2000
Y
BP-1000
BP-1000

Note: A JB-2 Junction Box may be used instead of the "Y" cables. The JB-2 provides a means of combining two channels, while also allowing the single channels to be continued directly from the junction box.

Figure 5 *- Typical connections using one SPS-2001 Power Supply for two intercom channels. The two program sources are monitored independently by their intercom channels. All audio (program and intercom) is monitored as a monaural mix in the SPS-2001 speaker. The US2000A is set for monaural speaker output (default).*



1 1-channel cable
2 2-channel cable
Y "Y" cable
SPS-2001
Power Supply
Combine / Isolate Switch set to Isolate
SPK1000
Powered Speaker
Speaker inter-connect cables.
1
1
US2000A
Master Station
Program input cable. From 2 audio sources
To all stations on channel 1
To all stations on channel 2
BP-1000
BP-1000
BP-1000
BP-1000
BP-2000
BP-2000
BP-1000
BP-1000
HEADSET
LINES
Y
2
Y

Note: A JB-2 Junction Box may be used instead of the "Y" cables. The JB-2 provides a means of combining two channels, while also allowing the single channels to be continued directly from the junction box.

Figure 6 *- Adding an SPK1000 to the example in Figure 5. The two program sources are monitored independently by the intercom channels. The SPS-2001 monitors intercom channel 1 and program 1. The SPK1000 monitors intercom channel 2 and program 2. The US2000A is set for binaural speaker output as described in the US2000A User Manual.*

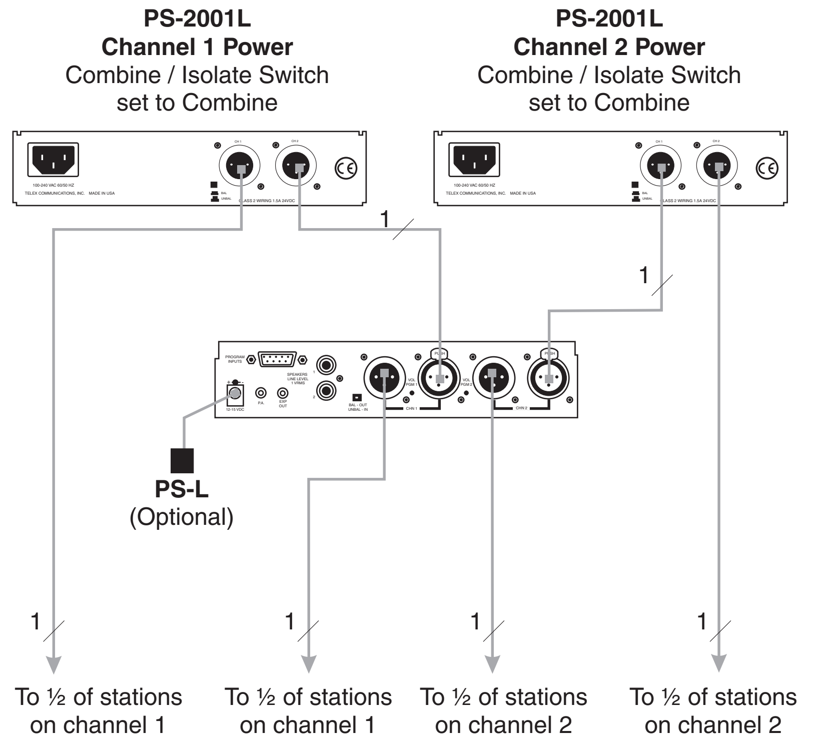

PS-2001L
Channel 1 Power
Combine / Isolate Switch
set to Combine
PS-2001L
Channel 2 Power
Combine / Isolate Switch
set to Combine
1
1
PS-L
(Optional)
1
1
1
1
To ½ of stations on channel 1
To ½ of stations on channel 1
To ½ of stations on channel 2
To ½ of stations on channel 2

Figure 7 *- Using two PS-2001L Power Supplies to provide a greater capacity on each channel.*

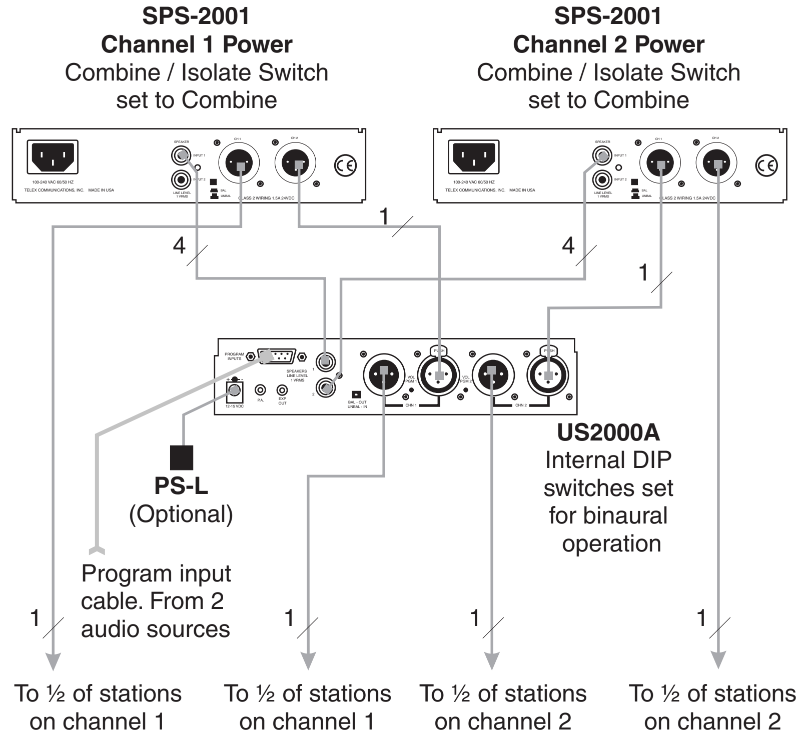

SPS-2001
Channel 1 Power
Combine / Isolate Switch
set to Combine
SPS-2001
Channel 2 Power
Combine / Isolate Switch
set to Combine
1
4
4
1
US2000A
Internal DIP switches set for binaural operation
PS-L
(Optional)
Program input cable. From 2 audio sources
1
1
1
1
To ½ of stations on channel 1
To ½ of stations on channel 1
To ½ of stations on channel 2
To ½ of stations on channel 2

Figure 8 *- Using two SPS-2001 Power Supplies to provide a greater capacity on each channel. Using two SPS-2001 Power Supplies also lets you independently monitor and adjust volume for both intercom channels without the need for a separate powered loudspeaker. In this application, the US2000A should be set for binaural speaker output as described int he US2000A manual.*

Power-Up Check

1. Plug in any DC wallpacks that are being used with individual intercom stations. Note: If you plug in DC wallpacks after applying power to the PS-2001L or SPS-2001, you may get an overload indication. This is because the stations that are being powered by wallpacks will draw current from the units until their own DC wallpack supplies are connected. To correct the problem, plug in the DC wallpacks.

2. Plug in the AC power cords for any connected PS-2001L or SPS-2001 Power Supplies. The channel status indicators should be green.

3. If a channel status indicator turns red during operation, locate the short or overload that is causing the problem. Once the short or overload is corrected, the Power Supply should reset itself and the overload indication should go away.

Specifications

General

Input Power Requirements: 100 to 240 VAC, 50/60 Hz
Output Power (each channel): 21±1 VDC, 2 A
Dimensions 1.75”(44.5 mm) high x 8.25” (209.5 mm) wide x 10.31” (261.9 mm) deep
Weight: approximately 2.5 lb (1.13 kg)
Environmental Requirements:
  Storage: -20°C to 80°C, 0% to 95% humidity, non-condensing
  Operating: 0°C to 50°C, 0% to 95% humidity, non-condensing

Intercom Channels

General

Connector Type: One XLR-3M audio connector for each channel. Pin-out depends on setting of BAL / UNBAL switch for balanced or unbalanced operation as defined below:

Balanced Mode (set to BAL position)

Line Terminating Impedance: 300 ohms ±10%
Connector Pin-out:

Pin 1	Common (audio and DC return)
Pin 2	Full-duplex, balanced intercom audio and +24 VDC output
Pin 3	Full-duplex, balanced intercom audio and +24 VDC output

Unbalanced Mode (set to UNBAL position)

Line Terminating Impedance: 150 ohms ±5%
Connector Pin-out:

Pin 1	Common (audio and DC return)
Pin 2	+24 VDC output
Pin 3	Full-duplex, unbalanced Intercom audio high

Speaker Inputs (SPS-2001 Only)

Type: RCA Phone Jack
  Tip: Speaker + Input
  Sleeve: Speaker - Input

Approvals

UL, CUL, CE

TRADEMARKS

Audiocom® is a registered trademark of Telex Communications. Names of other products mentioned herein are used for identification purposes only and may be trademarks and/or registered trademarks of their respective companies.

WARRANTY INFORMATION

Products are warranted by Telex Communications, Inc. to be free from defects in materials and workmanship for a period of one year from the date of sale.

The sole obligation of Telex during the warranty period is to provide, without charge, parts and labor necessary to remedy covered defects appearing in products returned prepaid to Telex. This warranty does not cover any defect, malfunction or failure caused beyond the control of Telex, including unreasonable or negligent operation, abuse, accident, failure to follow instructions in the manual, defective or improper associated equipment, attempts at modification and repair not authorized by Telex, and shipping damage. Products with their serial numbers removed or effaced are not covered by this warranty.

To obtain warranty service, follow the procedures entitled “Procedure for Returns” and “ Shipping to Manufacturer for Repair or Adjustment”.

This warranty is the sole and exclusive express warranty given with respect to Audiocom products. It is the responsibility of the user to determine before purchase that this product is suitable for the user’s intended purpose.

ANY AND ALL IMPLIED WARRANTIES, INCLUDING THE IMPLIED WARRANTY OF MERCHANTABILITY ARE LIMITED TO THE DURATION OF THIS EXPRESS LIMITED WARRANTY.

NEITHER TELEX NOR THE DEALER WHO SELLS TELEX PRODUCTS IS LIABLE FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES OF ANY KIND.

CUSTOMER SUPPORT

Technical questions should be directed to:

Customer Service Department
Telex
12000 Portland Avenue South
Burnsville, MN 55337 U.S.A
Telephone: (952) 884-4051
Fax: (952) 884-0043

RETURN SHIPPING INSTRUCTIONS

Procedure for Returns

If a return is necessary, contact the dealer where this unit was purchased.

If a return through the dealer is not possible, obtain a RETURN AUTHORIZATION from:

Customer Service Department
Telex Communications, Inc.
Telephone: 1-800-392-3497 or (952) 884-4051
Fax: 1-800-323-0498 or (952) 884-0043

DO NOT RETURN ANY EQUIPMENT DIRECTLY TO THE FACTORY WITHOUT FIRST OBTAINING A RETURN AUTHORIZATION.

Be prepared to provide the company name, address, phone number, a person to contact regarding the return, purchase order number, the type and quantity of equipment, a description of the problem and the serial number(s).

Shipping to Manufacturer for Repair or Adjustment

All shipments of products should be made via United Parcel Service or the best available shipper prepaid. The equipment should be shipped in the original packing carton; if that is not available, use any suitable container that is rigid and of adequate size. If a substitute container is used, the equipment should be wrapped in paper and surrounded with at least four inches of excelsior or similar shock-absorbing material. All returns must include the return authorization number. Units sent for repair or adjustment **DO NOT** need a return authorization number

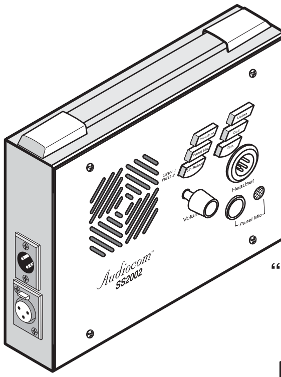
Factory Service department
Telex Communications, Inc.
West 1st Street
Blue Earth, MN 56013 U.S.A.

Upon completion of any repair the equipment will be returned via United Parcel Service or specified shipper collect.

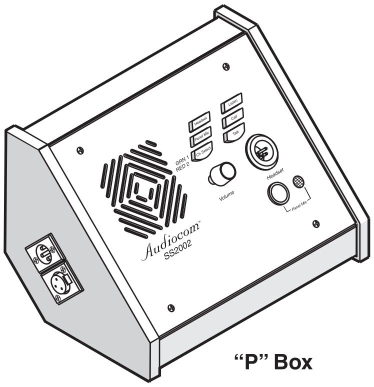
Telex®

User Instructions

Model SS-1002
Single Channel
Intercom Speaker
Station

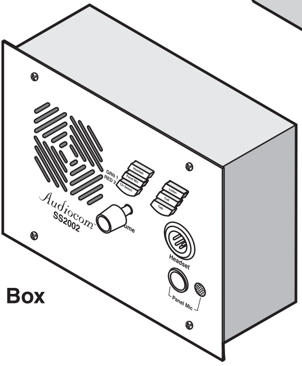

GRN 1
RED 2
Headset
Volume
Panel Mic
Audiocom
SS2002

"S" Box


GRN 1
RED 2
Headset
Volume
Panel Mic
Audiocom
SS2002

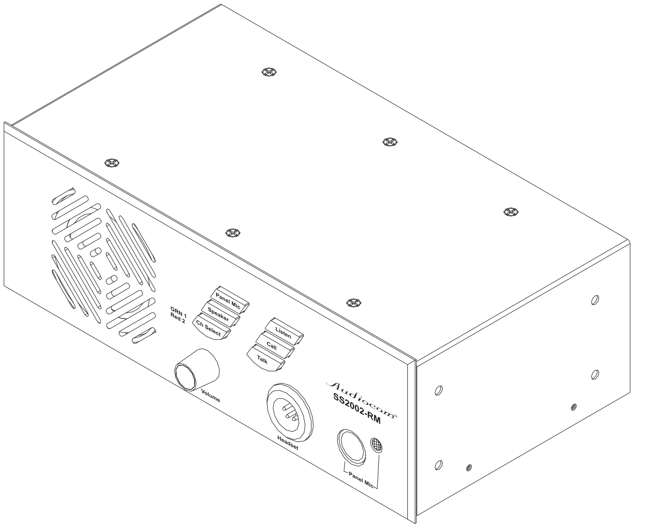
"P" Box

Model SS-2002
Two Channel
Intercom Speaker
Station


GRN 1
RED 2
Audiocom
SS2002
Headset
Panel Mic

"U" Box

Model SS-2002RM
Two Channel
Rack Mount
Intercom Speaker
Station


GRN 1
Red 2
Panel Mic
Speaker
Ch. Select
Listen
Call
Talk
Volume
Headset
Audiocom
SS2002-RM
Panel Mic

Audiocom® Intercom Systems

9350-7741-000 Rev A, 10/2002

FCC Statement

This equipment uses, and can radiate radio frequency energy that may cause interference to radio communications if not installed in accordance with this manual. The equipment has been tested and found to comply with the limits of a Class A computing device pursuant to Subpart J, Part 15 of FCC Rules which are designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area may cause interference which the user (at his own expense) will be required to correct.

CE This product meets Electromagnetic Compatibility Directive 89/336/EEC

Table of Contents

Introduction 4

Description 4

Features 4

Installation 6
Unpacking 6
Configuration Pre-check 6
DIP Switches 6
Power-Up 14
Sidetone Adjustment 14

Operation 15
Channel Select (SS-2002 & SS-2002RM Only) 15
Headset/Headphone/Speaker/Microphone Selection 15
Receiving Calls 15
Calling an Intercom Channel 15

Specifications 16
General 16
Dynamic-mic Headset/Handset 16
Panel Microphone 16
Intercom Channels, Balanced mode (SW1 set to BAL position) 16
Intercom Channels, Unbalanced mode (SW1 set to UNBAL position) 16
Speaker Output Power 17
Dimensions 17

Introduction

Thank you for purchasing the Audiocom SS-1002/SS-2002/SS-2002RM Intercom Speaker Station. We hope the many design features of this product will satisfy your intercommunication requirements for many years to come. To get the most out of your new intercom station, please take a few moments to look through this booklet before using the Intercom Speaker Station for the first time.

Description

The Intercom Speaker Stations may be used with a headset, or with the built-in speaker and the panel microphone or an optional gooseneck microphone. Another alternative is to use headphones and either the built-in panel microphone or an optional gooseneck microphone. As an alternative to a headset, a telephone style handset may also be used. The SS-1002 is a single-channel station; the SS-2002 & SS-2002RM provide switch selectable access to either of two intercom channels. Both the SS-1002 and SS-2002 come in three versions to suit a variety of applications. The “S” box is a portable version. It has a carrying handle and dual “loop-through” intercom connectors which permit stations to be quickly interconnected using prefabricated cables. The “P” box is a stationary version. It also has dual “loop-through” connectors for quick interconnection, but the case is designed for desktop or console-mount applications. The “U” box is designed for permanent, in-the-wall mounting. It uses push type wire terminals for connection to the intercom system. In addition, the SS-2002RM can be used on a desktop or mounted in a standard 19” equipment rack (using the optional RM-14 Rack Mount Kit).

Features

1. Channel and Power connections: The “U” box uses quick release terminals to connect audio and power wires. For the “S” box and “P” box versions, the terminal connections are brought out to the dual, loop-thru XLR connectors on the side of the box. These permit several stations to be connected in a string, or daisy-chain, using prefabricated intercom cables.
2. Configuration DIP switches: These switches control the following features:
   Call Beep: In addition to a flashing call key indication for incoming calls, an incoming call beep tone can also be used.
   Headset Microphone Type Selection: Either a balanced or unbalanced dynamic microphone can be selected.
   DC call enable: This may be turned on to use the intercom station with intercom systems that use DC call signaling.
3. Panel Mic Key: Selects the Panel Mic connector / Built-in Mic (11) in the on position and the headset connector (10) in the off position.
4. Speaker Key: Selects the built-in speaker in the on position and the Headset connector (10) in the off position.
5. Channel Select Key (SS-2002 & SS-2002RM only): Selects intercom channel 1 or 2. The key lights green for channel 1 and red for channel 2.
6. Volume Control: Adjusts intercom listen volume to headphones or speaker.
7. Intercom Listen Key: Both momentary (push-to-listen) and latching (hands-free listen) are possible.
8. Call Key: Used to send call signals on the intercom channel. Flashes for incoming calls.
9. Intercom Talk Key: Both momentary (push-to-talk) and latching (hands-free talk) are possible. All models are also equipped with a “mic kill receive” circuit, which allows an operator at a remote master station (such as the US-2000A) to turn off the talk key on all stations on the intercom line.
10. Dynamic-Mic Headset Connector: 4-pin male XLR connector accepts headsets with monaural headphones and either a balanced or unbalanced dynamic microphone. Also accepts a telephone style handset. Can also be used with headphones when a Panel Mic is connected. Can also be used with a handheld microphone along with the internal speaker.
11. Panel Mic Connector: Accepts an electret gooseneck microphone such as the Telex MCP-90.
12. Built-in Panel Microphone: Active when the Panel Mic key is pressed and a gooseneck microphone is not inserted in the Panel Mic connector.
13. Sidetone Trimmers: These adjust the level of the user’s own voice in the headphones when using full-cushion headphones. (Eliminates the muffled sensation when talking with the ears completely covered.) When using the speaker or open-ear style headphones, the sidetone trimmers are adjusted to eliminate the user’s voice from the headphones or speaker. This helps to prevent feedback.
14. Balanced / Unbalanced Selector Switch: This switch sets the intercom station for compatibility with either Audiocom (balanced) or Clear-Com* (unbalanced) intercom systems.

* Brand names mentioned are the property of their respective companies.

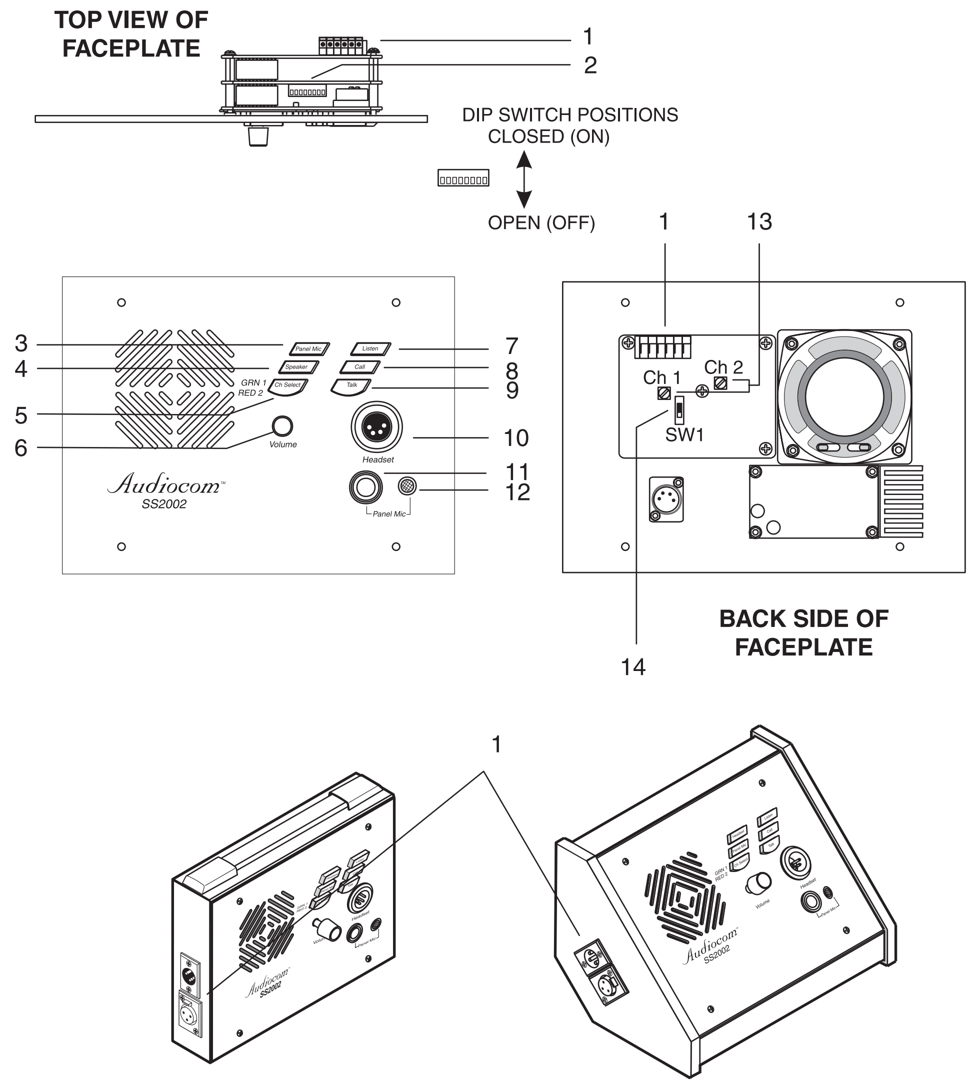

TOP VIEW OF FACEPLATE
1
2
DIP SWITCH POSITIONS
CLOSED (ON)
OPEN (OFF)
1
13
3
4
5
6
Panel Mic
Speaker
GRN 1
RED 2
Ch Select
Volume
Listen
Call
Talk
Headset
Panel Mic
7
8
9
10
11
12
Audiocom™
SS2002
Ch 1
Ch 2
SW1
14
BACK SIDE OF FACEPLATE
1

Figure 1 - SS-1002 / SS-2002 features.

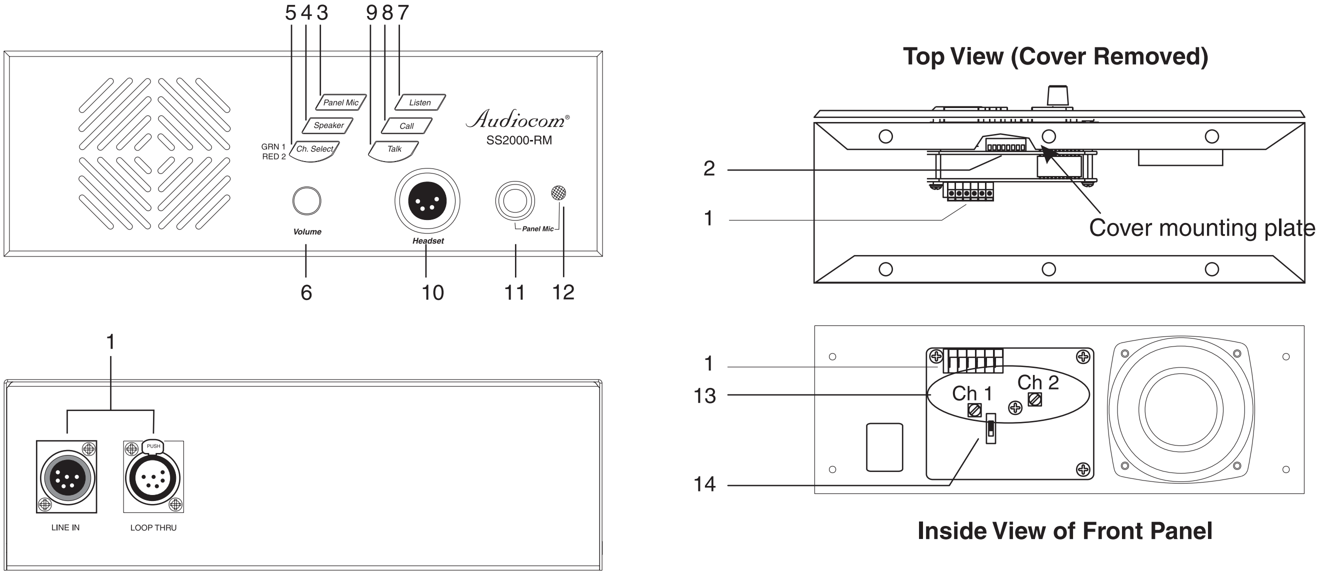

5 4 3
9 8 7
Panel Mic
Speaker
GRN 1
RED 2
Ch. Select
Listen
Call
Talk
Audiocom®
SS2000-RM
Volume
Headset
Panel Mic
6
10
11
12
1
LINE IN
PUSH
LOOP THRU
Top View (Cover Removed)
2
1
Cover mounting plate
1
13
Ch 1
Ch 2
14
Inside View of Front Panel

Figure 2 - SS-2002RM features.

Installation

Unpacking

Each SS-1002 / SS-2002 / SS-2002RM is supplied with the following items. Contact the shipper or your Audiocom dealer immediately if anything is damaged or missing. Be sure to fill out and return the product registration card to Telex to properly register your intercom station.

For SS-1002 / SS-2002 units:

Quantity	Description
1	SS1002 or SS2002 Intercom Station
1	Warranty and registration card
1	User Instructions

For SS-2002RM units:

Quantity	Description
1	SS2002RM Intercom Station
1	Warranty and registration card
1	User Instructions
4	Rubber feet (apply to bottom of SS2002RM for desktop use)

Configuration Pre-check

Before making connections, read the configuration notes that follow, and make sure that all internal controls are properly set for your intended usage. If you need to access the internal controls, remove the four screws on the front that secure the intercom station faceplate to the mounting box, then lift out the faceplate for all versions except the RM version. To access the internal controls for the RM version, remove the 10 screws securing the cover on the box and then lift the cover.

DIP Switches

Table 1 lists the DIP switch descriptions and factory default settings.

Switch Number	**Description**	**Settings** Open=Off, Closed=On	**Default** Setting
1	Mic kill receive	Closed: Disabled-no mic kill Open: Enabled-mic kill active	Open
2	Call signal method	Closed: DC (SW1 set to UNBAL) Open: Audiocom (SW1 set to BAL)	Open
3	Incoming call beep	Closed: Disabled Open: Enabled	Open
4	Microphone type	Closed: Unbalanced Open: Balanced	Open
5	Speaker beep for incoming call	Closed: Enabled (DIP switch 3 must be set to Open) Open: Disabled	Open
6	Not used	Don't care	Open
7	Not used	Don't care	Open
8	Not used	Don't care	Open

Mic Kill Receive

Audiocom master stations, such as the US-2002A, can transmit an inaudible signal to turn off the microphones in all remote intercom stations on an intercom channel. This is useful when a remote intercom station has been left unattended with the microphone on. The procedure to send a mic kill signal from a master station is a two-step process, so that it is very unlikely that microphones will ever be turned off by accident. However, you may wish to disable the mic kill receive feature at the intercom unit. You might do this, for example, if communications will be of a very critical nature where it is absolutely essential the microphone never be remotely deactivated. To enable the mic kill receive function set DIP switch 1 to the open position. To disable the mic kill function set DIP switch 1 to the closed position.

Call Signal Compatibility

For Audiocom setup, leave this switch in the open position, and also leave the Balanced / Unbalanced switch in the Balanced position. For Clear-Com setup, set DIP switch 2 to the closed position and set the Balanced / Unbalanced switch in the Unbalanced position.

Incoming Call Beep

Incoming calls are always indicated by a red-flashing call key. If you also want incoming call beep in headphones, set DIP switch 3 to the Open position (default). DIP switch 3 must also be in the Open position if you want to activate incoming call beep to the speaker using DIP switch 5.

Microphone Type

If you will be using a headset, telephone-style handset, or dynamic handheld microphone, and the specifications indicate the microphone type is balanced, leave DIP switch 4 in the Open (default) position. If the specifications indicate an unbalanced microphone set this switch to the Closed position. If you are not sure, leave the switch in the Open position for now, then try both positions during the operation. Use the position that produces the best audio quality at other stations.

Speaker Beep for Incoming Call

Set DIP switch 5 to the Closed position if you want to hear incoming call beep in the speaker. Note that DIP switch 3 must be in the Open position.

Balanced/Unbalanced Switch

This switch is set at the factory to the Balanced, or BAL, position for use with Audiocom intercom channels. Set the switch to the Unbalanced, or UNBAL position for use with a Clear-Com intercom system.

Sidetone Trimmers

These are normally adjusted after all connections are completed. See “Sidetone Adjustment.”

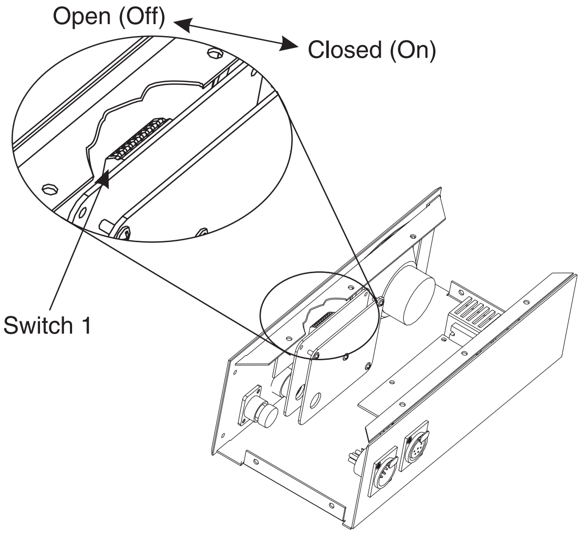


Figure 3 - DIP Switch Location RM Version.

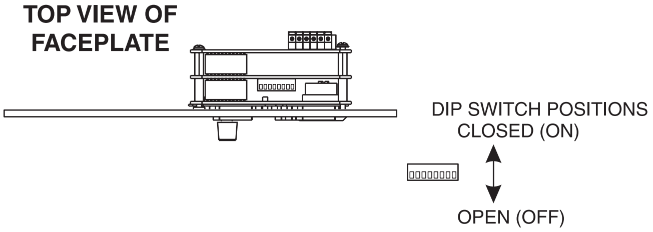


Figure 4 - DIP Switch Location S, U, and P Versions.

Headset Connector Notes

If you are using a monaural, dynamic-mic headset, or a monaural, telephone-style, dynamic-mic handset, or a handheld dynamic microphone, plug it into the Headset connector. You can also use this connector with monaural headphones for listening when you are using either the built in panel microphone, or the optional gooseneck microphone for talk back.

Panel Mic Connector Notes

As an alternative to a headset or telephone-style handset, you can use either the built-in panel microphone or an optional gooseneck microphone. To connect a Telex MCP-90 gooseneck microphone, insert the 1/4” TRS connector into the panel microphone connector and screw the microphone in. When a gooseneck microphone is inserted in the panel microphone connector, the built-in panel microphone is automatically bypassed. Use either of these microphones for talk back, then use the internal speaker for listening, or connect a pair of headphones to the headset connector for private listening.

Intercom Channel Connections

the method of connection depends on the type of box. The “S” box, “P” box, and RM versions connect to the intercom system using prefabricated cables in a phantom-powered configuration. The “U” box version is designed for permanent installation, and it can be either phantom powered or locally powered.

Description of Phantom Powered Connection

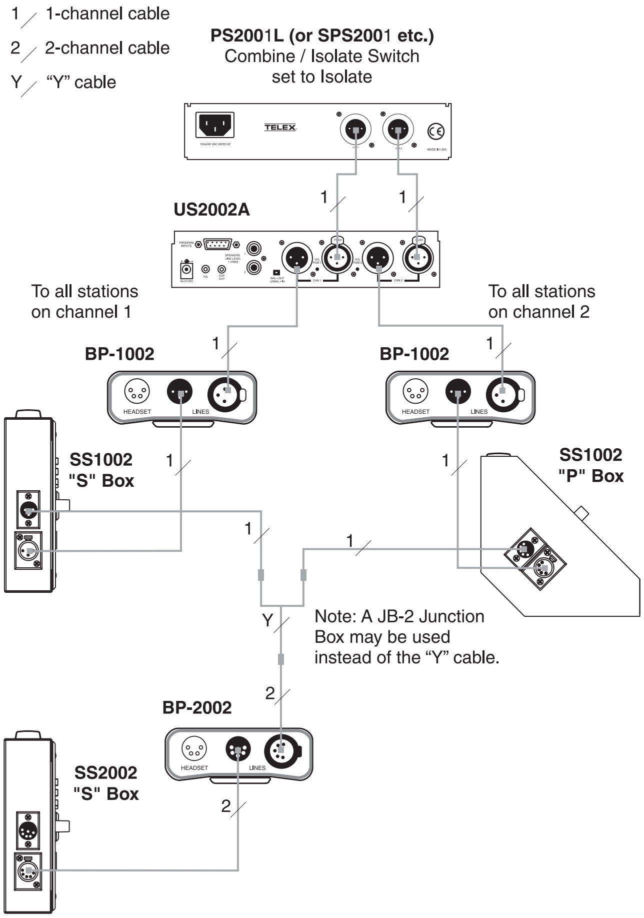

1 1-channel cable
2 2-channel cable
Y “Y” cable
PS2001L (or SPS2001 etc.)
Combine / Isolate Switch
set to Isolate
TELEX
US2002A
1
1
To all stations
on channel 1
To all stations
on channel 2
BP-1002
BP-1002
HEADSET
LINES
SS1002
"S" Box
SS1002
"P" Box
Y
Note: A JB-2 Junction
Box may be used
instead of the “Y” cable.
2
BP-2002
SS2002
"S" Box

Figure 5 - An example of a phantom powered system. In this system a PS-2001L power supply is set to isolate mode. In this mode each channel is a separate party line and current per channel is limited to 1 amp. Note that both single and two channel intercom stations may be connected by using “Y” cables (or JB2 Junction boxes). Prefabricated cables may be used, or cables may be constructed using the diagrams in FIgure 11.

In this method, operating power and intercom audio for the entire intercom system ware carried over the same wires. The advantage of this method is simplicity of connection. The system power supply (PS-2001L, SPS-2001A) etc.) automatically provides what is know as a terminating impedance for the intercom system. Without this terminating impedance, the sound quality will be distorted, and the levels will shift every time additional stations are connected. The disadvantage of the phantom power method is that some operating power is lost over very long intercom cables, and performance at distant intercom stations may be reduced. Generally, if intercom stations are within a few hundred feet from the system power supply, phantom power will be sufficient. Also note that increasing the number of stations will reduce the overall operating distance. Phantom power is generally the only method that will be used to connect the “S” box, “P” box, and RM versions. Figure 5 illustrates a phantom powered intercom system.

Note that the distance over which power can be delivered is less than the distance over which audio can be delivered. Audio can be delivered for several miles to locally powered stations as described below.

Description of Locally Powered Connection

Using this method, an intercom station is connected to the intercom line just like any phantom-powered intercom station, except that a local power supply is also connected. This local power supply is located with the intercom station and provides power for that station only. Since power loss on the intercom lines is no longer an issue, the operating range is now limited only to the audio transmission range, which is several miles. Another advantage to this method is that more stations can be connected to the intercom channels. When local power is supplied to an intercom station, the station detects this and automatically disconnects from the system’s phantom power supply.

Figure 6 illustrates an intercom system with both phantom powered and locally powered “U” boxes. As long as a system power supply (PS-2001L, SPS-2001, etc.) is located somewhere in the intercom system, the proper terminating impedance will still be suppled for all stations. However, if all stations are locally powered, and there is not system power supply, a special line termination must be installed.

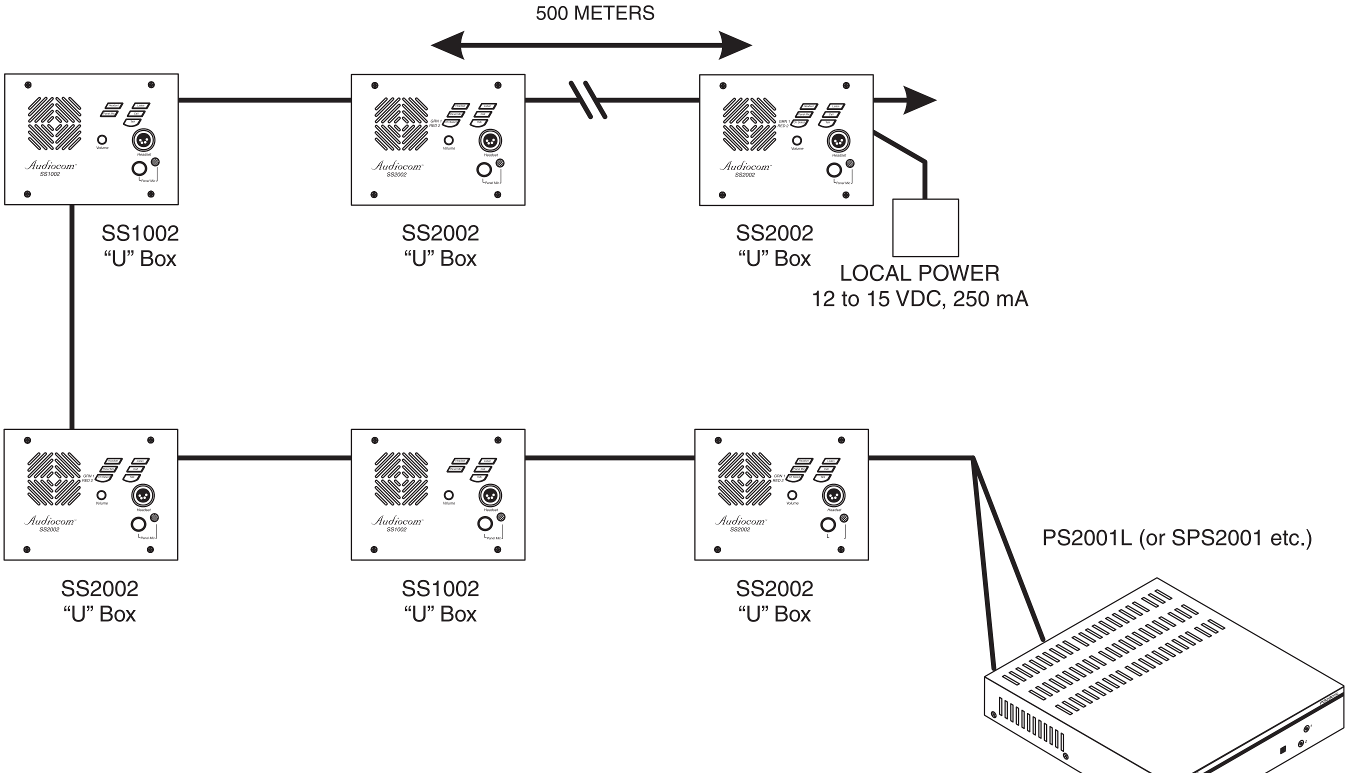


Figure 6 - A two-channel Audiocom intercom system using a PS2001L power supply. the PS2001L is set to isolate mode. In isolate mode, each intercom channel is a separate party line, and total current for each channel is limited to 1 amp. Note that both SS1002 and SS2002 stations may be connected depending on each location’s need to communicate with one or two channels. SS1002 stations may be connected to either channel one or two. Also note that locally powered stations may be connected. This is recommended when stations are installed at remote locations. Since the PS2001L provides termination for the intercom channels, no user-installed termination is required.

All Locally Powered Intercom Stations (Dry Lines)

If intercom stations are widely distributed, you can dispense with a system power supply (PS2001L, SPS-2001, etc.) and use local power for each station. When no power is delivered on the intercom channels, this is known as dry-line operation. Since a system power supply is not used, a line termination must be inserted in each intercom channel for proper operation. Again, note that only the “U” box has provisions for local power. The 3-pin connectors on the “S”, “P”, and RM versions do not permit this type of connection. Figure 7 illustrates an intercom system using all locally powered “U” boxes.

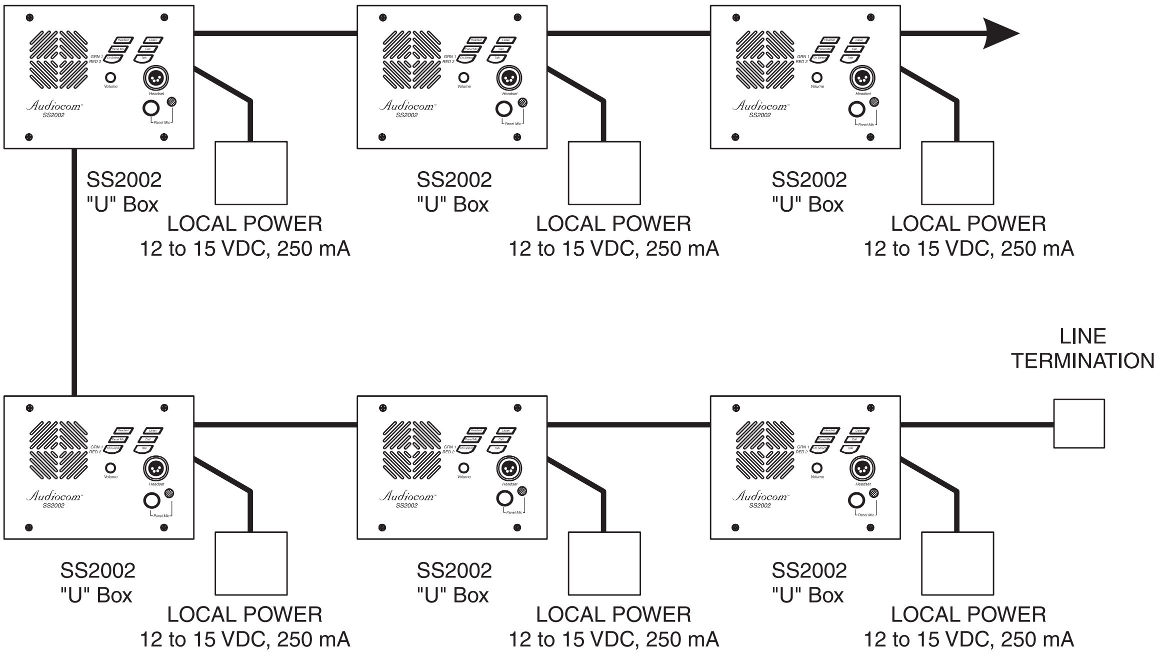

SS2002
"U" Box
LOCAL POWER
12 to 15 VDC, 250 mA
SS2002
"U" Box
LOCAL POWER
12 to 15 VDC, 250 mA
SS2002
"U" Box
LOCAL POWER
12 to 15 VDC, 250 mA
LINE
TERMINATION
SS2002
"U" Box
LOCAL POWER
12 to 15 VDC, 250 mA
SS2002
"U" Box
LOCAL POWER
12 to 15 VDC, 250 mA
SS2002
"U" Box
LOCAL POWER
12 to 15 VDC, 250 mA

Figure 7 - An example of an Audicom intercom system using all locally powered “U” box stations, with no power being distributed on the intercom channels (dry lines). Since an Audiocom power supply is not used, the installer must connect a line termination somewhere in each channel for proper operation.

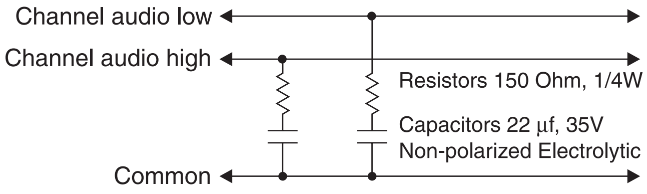
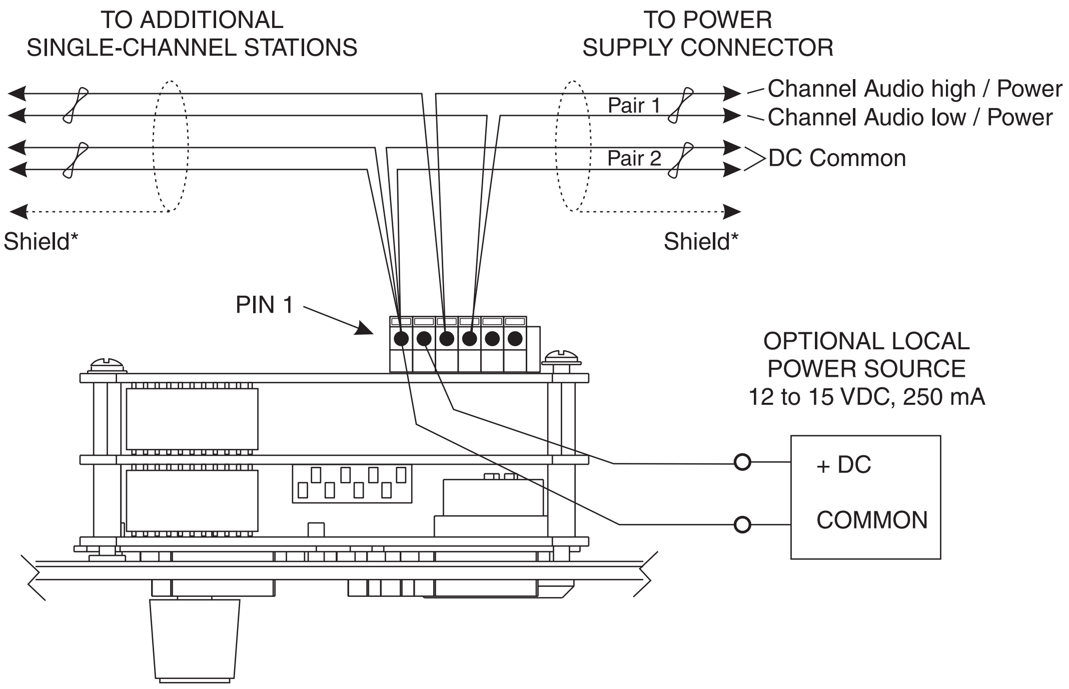

Channel audio low
Channel audio high
Resistors 150 Ohm, 1/4W
Capacitors 22 µf, 35V
Non-polarized Electrolytic
Common

Figure 8 - Audiocom mode line termination for dry-line operation. (One required for each channel). the termination is available via PN: 9020-7550-000.


TO ADDITIONAL
SINGLE-CHANNEL STATIONS
TO POWER
SUPPLY CONNECTOR
Pair 1
Channel Audio high / Power
Channel Audio low / Power
Pair 2
DC Common
Shield*
Shield*
PIN 1
OPTIONAL LOCAL
POWER SOURCE
12 to 15 VDC, 250 mA
+ DC
COMMON

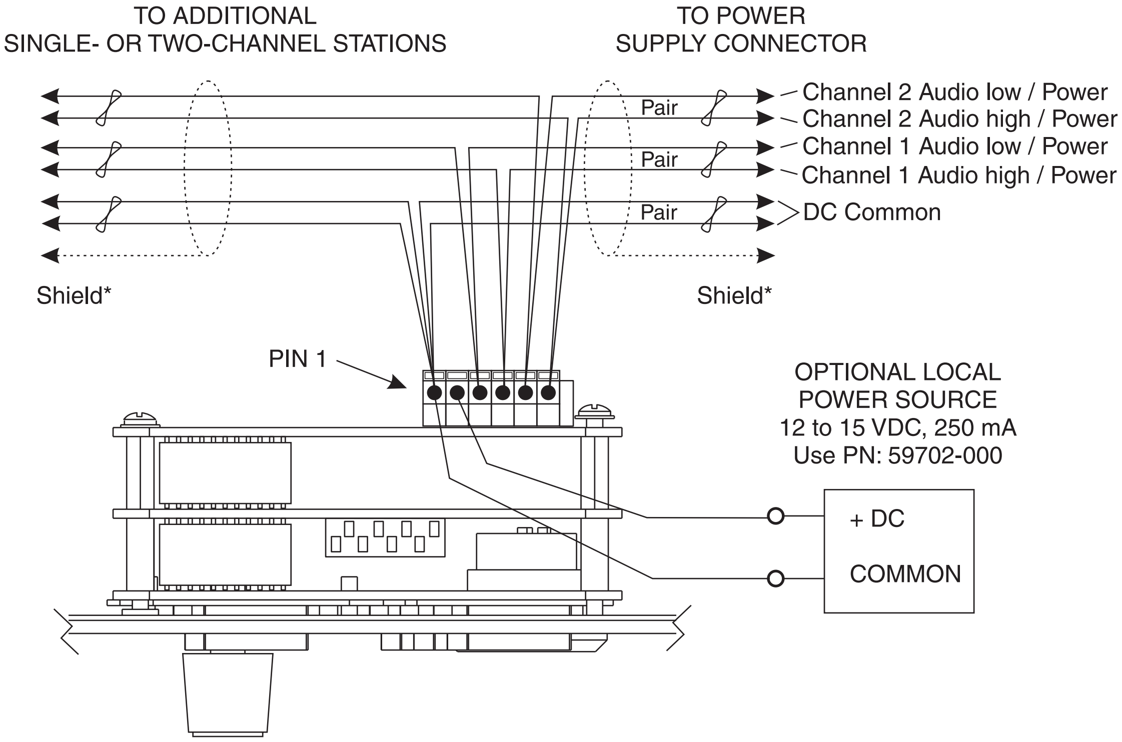
Cable Type: 22AWG Stranded, 2-Pair Twisted-wire, with Shield

Denotes twisted pair.

Denotes shield.

*Shield: Earth ground (Connect at power supply only. Do not short to DC common)

Figure 9 - Audiocom mode connections for an SS1002 Intercom Station in a “U” box. Note: for Clear-Com connection, use the Unbalanced Mode Intercom Channel pin-out information listed in the Specifications section.


TO ADDITIONAL
SINGLE- OR TWO-CHANNEL STATIONS
TO POWER
SUPPLY CONNECTOR
Pair
Pair
Pair
Channel 2 Audio low / Power
Channel 2 Audio high / Power
Channel 1 Audio low / Power
Channel 1 Audio high / Power
DC Common
Shield*
Shield*
PIN 1
OPTIONAL LOCAL
POWER SOURCE
12 to 15 VDC, 250 mA
Use PN: 59702-000
+ DC
COMMON

Cable Type: 22AWG Stranded, 3-Pair Twisted-wire, with Shield

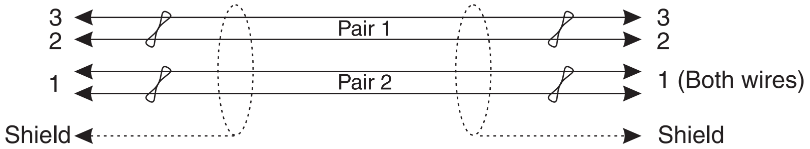
Denotes twisted pair.

Denotes shield.

*Shield: Earth ground (Connect at power supply only. Do not short to DC common)

Figure 10 - Audiocom mode connections for an SS2002 Intercom Station in a "U" box. Note: for Clear-Com connection, use the Unbalanced Mode Intercom Channel pin-out information listed in the Specifications section.

TYPICAL 1-CHANNEL CABLE WIRING


3
2
1
Shield
Pair 1
Pair 2
3
2
1 (Both wires)
Shield

Cable Type: 22AWG Stranded, 2-Pair Twisted-wire, with Shield

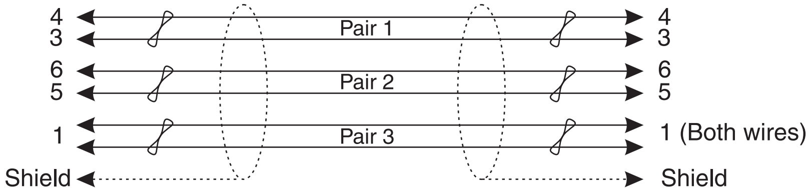
Connector Type: 3-Pin XLR Audio (Neutrik or Switchcraft)*

Pin 1: Common
Pin 2: Channel Audio / Power
Pin 3: Channel Audio / Power
Shield: Earth ground

Denotes twisted pair.

Denotes shield.

TYPICAL 2-CHANNEL CABLE WIRING


4
3
6
5
1
Shield
Pair 1
Pair 2
Pair 3
4
3
6
5
1 (Both wires)
Shield

Cable Type: 22AWG Stranded, 3-Pair Twisted-wire, with Shield

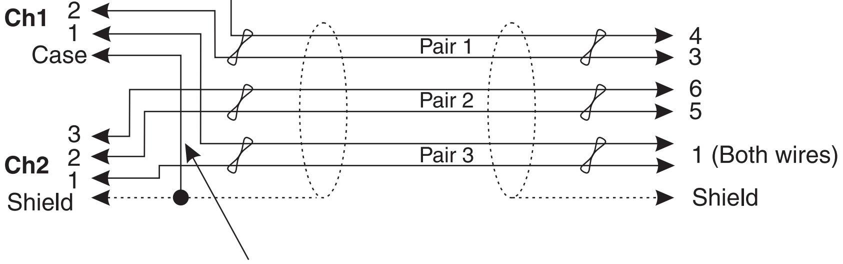
Connector Type: 6-Pin XLR Audio (Neutrik only, not compatible with 6-pin Switchcraft)*

Pin 1: Channel 1 & 2 Common
Pin 2: No connection
Pin 3: Channel 1 Audio / Power
Pin 4: Channel 1 Audio / Power
Pin 5: Channel 2 Audio / Power
Pin 6: Channel 2 Audio / Power
Shield: Earth ground

Denotes twisted pair.

Denotes shield.

“Y” CABLE WIRING


Ch1
3
2
1
Case
Ch2
3
2
1
Shield
Pair 1
Pair 2
Pair 3
4
3
6
5
1 (Both wires)
Shield

Use second drain wire if available, or add an extra section of wire.

* Standard cables are generally constructed using a male connector at one end and a female connector at the other end. This allows several cables to be interconnected to create longer cable runs. Audiocom master stations, speaker stations and belt packs also typically provide both a male and female Neutrik connector for each intercom channel. This permits loop-through connection of several intercom stations using the standard cables. Audiocom power supplies use a 3-pin male Neutrik connector for each channel. Audiocom wallplates use male Neutrik connectors.

Figure 11 - Audiocom cable wiring diagrams.

Power-Up

Make sure any local power supplies are plugged in, then turn on the power switches of any phantom power supplies (PS-2001L, SPS-2001, etc.)

Sidetone Adjustment

The SS-1002, SS-2002, and SS-2002RM use full-duplex audio (the same as conventional telephone lines) in which the talk and listen audio are sent and received on the same wires. Thus, when you talk, you also here your own voice back as with a telephone. If you are using the built-in speaker for listening and a either a panel microphone or handheld microphone for talkback, you could experience unwanted feedback since the microphone may pick up your returned voice audio from the speaker and re-amplify it. This could also happen if you are using a headset or headphones with open-ear style earmuffs. On the other hand, if you are using earmuffs that completely enclose the ears, the microphone is not likely to pick up the returned audio. A certain amount of your own voice level is desirable to overcome the muffled sensation when talking. The sidetone adjustment is different for these different operating conditions:

If you are suing the speaker with a panel microphone, or if you are using an open-ear style headset or headphones, adjust sidetone as follows:

1. SS-2002 & SS-2002RM only: activate channel 1 as described in the operating instructions.

2. Activate talk and listen as described in the operating instructions

3. Slowly increase the volume to maximum while talking into the microphone.

4. Using a small flat-bladed screwdriver, adjust the channel 1 sidetone trimmer to minimize your voice level in the headphones.

5. SS-2002 & SS-2002RM only: activate channel 2 and repeat the above steps to adjust the channel 2 sidetone.

6. Install the intercom station mounting screws after completing the adjustments. The station is now ready for use.

If you are using headphones that completely enclose the ears, adjust sidetone as follows:

1. SS-2002 & SS-2002RM only: activate channel 1 as described in the operating instructions.

2. Activate talk and listen as described in the operating instructions.

3. Set the volume control to the normal listening level for intercom audio

4. While talking into the microphone, use a small flat-bladed screwdriver to adjust the channel 1 sidetone trimmer so that you can hear your own voice in the headphones at an acceptable level.

5. SS-2002 & SS-2002RM only: activate channel 2 and repeat the above steps to adjust the channel 2 sidetone.

Operation

Channel Select (SS-2002 & SS-2002RM Only)

Tap the Ch Select key to select channel 1 or 2. The key is green when channel 1 is selected and red when channel two is selected.

Headset/Headphone/Speaker/Microphone Selection

- To use a headset or telephone style handset, set the Speaker and Panel Mic keys to off.
- To use headphones with the built-in panel microphone, set the Speaker key to off, set the Panel Mic Key to on, and make sure that a gooseneck microphone is not inserted in the panel microphone connector.
- To use headphones with a gooseneck microphone, set the Speaker key to off, set the Panel Mic Key to on, and make sure that a gooseneck microphone is inserted in the panel microphone connector.
- To use the speaker with a handheld dynamic microphone, set the speaker key to on and set the Panel Mic Key to off.
- To use the speaker with the built-in panel microphone, set the Speaker and Panel Mic keys to on, and make sure that a gooseneck microphone is not inserted in the panel microphone connector.
- To use the speaker with a gooseneck microphone, set the Speaker and Panel Mic keys to on, and make sure that a gooseneck microphone is inserted in the panel microphone connector.

Receiving Calls

When there is an incoming call signal, the Call key will flash red. There will also be a beep tone in the headphones or speaker if the beep feature is activated.

NOTE: The incoming call indication is provided only for the currently selected intercom channel in the case of SS-2002 & SS-2002RM models.

Turn on the Talk and Listen keys and begin your conversation. Turn the keys off when finished.

NOTE: You can turn the talk and listen keys on in either momentary or latched mode. For momentary operation, press and hold the key. For latched operation, tap the key to turn it on. Then tap it again to turn it off when finished.

Calling an Intercom Channel

1. SS-2002 & SS-2002RM only: select the desired intercom channel.

2. Press and hold the Call key. An inaudible call signal will be sent, and the Listen key will automatically turn on.

3. When you hear a response, release the call key and activate the talk key.

4. Turn off your Talk and Listen keys to end the conversation.

Specifications

General

Power Requirements:

Phantom Power: +24 VDC nominal, 175 mA
Local Power: +12 to +15 VDC, 250 mA

Dimensions: See dimension drawings.

Environmental Requirements:

Storage: -20°C to 80°C; 0% to 95% RH, non-condensing
Operating: -15°C to 60°C; 0% to 95% RH, non-condensing

Dynamic-mic Headset/Handset

Microphone: 50 to 200 ohms, dynamic (balanced or unbalanced)
Headphones: 150 to 600 ohms, monaural
Connector Type: XLR-4M

Pin 1 - Microphone Low
Pin 2 - Microhpne High
Pin 3 - Headphone High
Pin 4 - Headphone Low

Panel Microphone

Built-in Microphone:

2.2k ohms, electret (-44 dB nominal), Omnidirectional

Gooseneck Microphone:

Connector Type: ¼” Threaded TRS
5k ohms, electret (-47 dB nominal)
Tip - Microphone High (+12 VDC bias)
Ring - Common
Sleeve - Common
Compatible with MCP-90 Series Microphones

Intercom Channels, Balanced mode (SW1 set to BAL position)

Output Level: 1Vrms nominal
Input Impedance: 300 ohms
Bridging Impedance: greater than 10k ohms
Sidetone: -40 dB, 35 dB adjustable range
Call Signaling:

Send: 20 kHz ±100 Hz, 0.5 Vrms ±10%
Receive: 20 kHz ±800 Hz, 100 mVrms

Mic-Kill Detect Frequency: 24 kHz ±800 Hz, 100mVrms
Noise Contribution: less than -70 dB
Common Mode Rejection Ratio: greater than 50 dB
Connector Type: Six-position push type terminal block
Pin 1 - Audio and DC Common
Pin 2 - Local power (+12 to +15 VDC) If needed
Pin 3 - Intercom channel 1 audio low and +24 VDC power
Pin 4 - Intercom channel 1 audio high and +24 VDC power
Pin 5 - Intercom channel 2 audio low and +24 VDC power
Pin 6 - Intercom channel 2 audio high and +24VDC power

Intercom Channels, Unbalanced mode (SW1 set to UNBAL position)

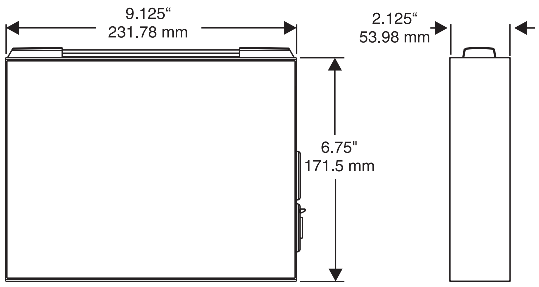
Output Level: 0.70 Vrms ±10%
Input Impedance: 200 ohms
Bridging Impedance: greater than 10k ohms
Sidetone: -40 dB, 35 dB adjustable range

Call Signaling:
Pin 1 - Common
Pin 2 - Local power (+12 to +15 VDC) If needed
Pin 3 - Power +30 VDC
Pin 4 - Intercom channel 1 audio and DC call signaling
Pin 5 - Power +30 VDC
Pin 6 - Intercom channel 2 audio and DC call signaling

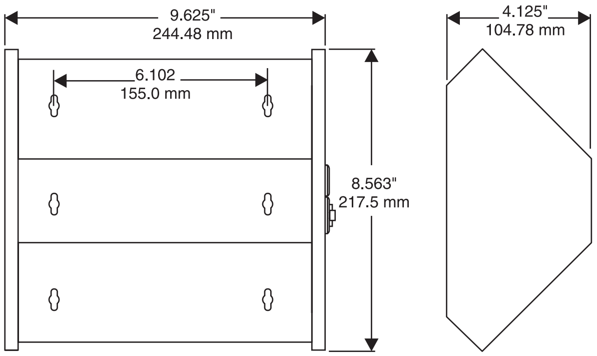
Speaker Output Power

4W max @ 8 ohms

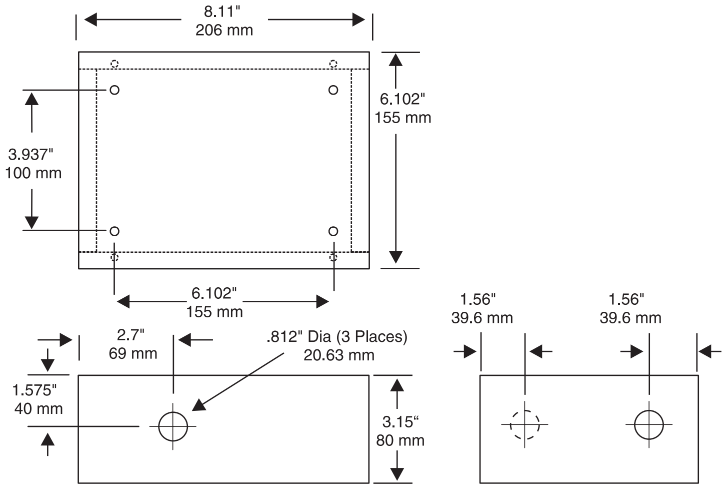
Dimensions


9.125“
231.78 mm
2.125“
53.98 mm
6.75"
171.5 mm

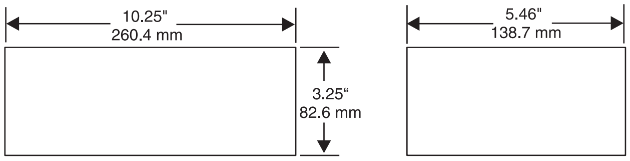
"S" Box


9.625"
244.48 mm
6.102
155.0 mm
8.563"
217.5 mm
4.125"
104.78 mm

"P" Box


8.11"
206 mm
6.102"
155 mm
3.937"
100 mm
6.102"
155 mm
2.7"
69 mm
.812" Dia (3 Places)
20.63 mm
1.575"
40 mm
3.15“
80 mm
1.56"
39.6 mm
1.56"
39.6 mm

"U" Box


10.25"
260.4 mm
3.25“
82.6 mm
5.46"
138.7 mm

RM

TRADEMARKS

Audiocom® is a registered trademark of Telex Communications. Names of other products mentioned herein are used for identification purposes only and may be trademarks and/or registered trademarks of their respective companies.

WARRANTY INFORMATION

Products are warranted by Telex Communications, Inc. to be free from defects in materials and workmanship for a period of one year from the date of sale.

The sole obligation of Telex during the warranty period is to provide, without charge, parts and labor necessary to remedy covered defects appearing in products returned prepaid to Telex. This warranty does not cover any defect, malfunction or failure caused beyond the control of Telex, including unreasonable or negligent operation, abuse, accident, failure to follow instructions in the manual, defective or improper associated equipment, attempts at modification and repair not authorized by Telex, and shipping damage. Products with their serial numbers removed or effaced are not covered by this warranty.

To obtain warranty service, follow the procedures entitled “Procedure for Returns” and “ Shipping to Manufacturer for Repair or Adjustment”.

This warranty is the sole and exclusive express warranty given with respect to Audiocom products. It is the responsibility of the user to determine before purchase that this product is suitable for the user’s intended purpose.

ANY AND ALL IMPLIED WARRANTIES, INCLUDING THE IMPLIED WARRANTY OF MERCHANTABILITY ARE LIMITED TO THE DURATION OF THIS EXPRESS LIMITED WARRANTY.

NEITHER TELEX NOR THE DEALER WHO SELLS TELEX PRODUCTS IS LIABLE FOR INCIDENTAL OR CONSEQUENTIAL DAMAGES OF ANY KIND.

CUSTOMER SUPPORT

Technical questions should be directed to:

Customer Service Department
Telex
12000 Portland Avenue South
Burnsville, MN 55337 U.S.A
Telephone: (952) 884-4051
Fax: (952) 884-0043

RETURN SHIPPING INSTRUCTIONS

Procedure for Returns

If a return is necessary, contact the dealer where this unit was purchased.

If a return through the dealer is not possible, obtain a RETURN AUTHORIZATION from:

Customer Service Department
Telex Communications, Inc.
Telephone: 1-800-392-3497 or (952) 884-4051
Fax: 1-800-323-0498 or (952) 884-0043

DO NOT RETURN ANY EQUIPMENT DIRECTLY TO THE FACTORY WITHOUT FIRST OBTAINING A RETURN AUTHORIZATION.

Be prepared to provide the company name, address, phone number, a person to contact regarding the return, purchase order number, the type and quantity of equipment, a description of the problem and the serial number(s).

Shipping to Manufacturer for Repair or Adjustment

All shipments of products should be made via United Parcel Service or the best available shipper prepaid. The equipment should be shipped in the original packing carton; if that is not available, use any suitable container that is rigid and of adequate size. If a substitute container is used, the equipment should be wrapped in paper and surrounded with at least four inches of excelsior or similar shock-absorbing material. All returns must include the return authorization number. Units sent for repair or adjustment **DO NOT** need a return authorization number

Factory Service department
Telex Communications, Inc.
West 1st Street
Blue Earth, MN 56013 U.S.A.

Upon completion of any repair the equipment will be returned via United Parcel Service or specified shipper collect.